APPENDIX D

TEST OBJECTIVES, CRITERIA, AND DATA REQUIRED

D-1 OVERVIEW. The following sections outline the test objectives, criteria and data required for Joint Interoperability Certification (JIC) of Defense Switched Network (DSN) switching systems. Requirements are annotated as to the applicability per switch type: Tandem Switch (TS), Multifunction Switch (MFS), End Office Switch (EOS), Small End Office (SMEO), Private Branch Exchange Type 1 (PBX1), and Private Branch Exchange Type 2 (PBX2). Remote Switching Units (RSUs) must meet all appropriate switch type procedures (i.e., EOS, SMEO or PBX) and section E-4. To be certified for Voice over Internet Protocol (VoIP), systems must meet all applicable switch type requirements and meet section E-5. Detailed test procedures for each of the sections below are contained in appendix E.

D-2 DSN INTERFACES

D-2.1 2-Wire Analog Line Interface

[Required: MFS, EOS, SMEO, PBX1]

D-2.1.1 Access

Objective. To determine System Under Test's (SUT's) ability to meet Generic Switching Center Requirements (GSCR) line requirements for a 2-Wire analog interface for: Directory Number (DN) identification, Direct Inward Dial (DID), line signaling, alerting signals and tones, and Worldwide Numbering and Dialing Plan (WWNDP).

Criteria

a. DN. SUT must be capable of automatically identifying the DN.

b. Line signaling. SUT shall support:

1) Loop start line.

2) Ground start line.

c. Alerting signals and tones. SUT must provide:

1) Ringing.

2) DSN Information Signals. Ringback precedence call, preemption tone, call waiting, conference disconnect tone, override tone, and camp on.

d. WWNDP. SUT must support the WWNDP format.

1) DSN user dialing consisting of:

a) Access digit N, where N is any digit 2-9.

b) Precedence (P) or service (S) digit, where P is any digit 0-4 and S is any digit 5-9.

c) Route code 1X, where X is any digit 0-9.

d) Area code KXX, where X is any digit 0-9.

e) Switch code KXX, where X is any digit 0-9.

f) Line number XXXX, where X is any digit 0-9.

2) 7-digit/10-digit intraswitch dialing.

3) 911 Conflict Resolution.

e. Call treatments. SUT shall provide following call treatments:

1) Origination busy treatment.

2) Busy/Idle Status treatment.

f. Class of Service.

1) SUT Shall provide a minimum of 256 classmarks.

g. Screening. SUT shall provide the following screening IAW GSCR Section 4.5.8:

1) Zone restriction capacity.

2) Access restriction.

3) COS Screening.

4) Zone Restriction

Test Procedures. See detailed test procedures in section E-2.1.1 of appendix E.

Data Required

a. SUT configuration.

b. Alerting tones/signals.

c. Dialed digits.

d. Call processing.

e. Call treatments.

D-2.1.2 Voice

Objective. To determine the SUT's capability to transport DSN Voice services, non-secure and secure, via a 2-wire analog interface.

Criteria

a. Voice quality measured from end instrument to end instrument across the SUT must have a Mean Opinion Score (MOS) of 4.0 95% of the time for non-secure voice.

b. Multi-Level Precedence and Preemption (MLPP) shall be IAW GSCR Sections 3. SUT shall meet MLPP interaction and required tones for:

1) Precedence levels.

2) Invocation and Operation.

c. DSN Announcements. SUT shall support announcements per GSCR Table 3-1.

d. MLPP Community of Interest (COI). SUT shall support COI/MLPP interaction.

e. MLPP COI Precedence Treatment. SUT shall support COI treatments.

f. SUT shall be able to complete Secure calls via a Secure Terminal Unit 3rd Generation (STU-III), Secure Terminal Equipment (STE), or Secure Wire Line Terminal (SWT). Secure calls shall complete at a minimum of 4.8 kilobits per second (kbps).

Test Procedures. See detailed test procedures in section E-2.1.2 of appendix E.

Data Required. During the test conduct collect the following data:

a. MOS.

b. Secure call completion rates.

c. Secure call data rates.

d. MLPP signaling tones.

e. DSN Announcements.

f. COI.

D-2.1.3 Facsimile

Objective. To determine SUT's ability to transport DSN facsimile traffic across a 2-wire analog interface.

Criteria. Facsimile traffic shall be able to be completed across the SUT with Group 3 facsimile rates of 14.4 kbps. Completed facsimiles shall meet Institute of Electrical and Electronics Engineers (IEEE) Standard (Std.) 167A-1995 facsimile readability requirements.

Test Procedures. See detailed test procedures in section E-2.1.3 of appendix E.

Data Required. Completed facsimile IEEE Std. 167A-1995 (Facsimile Test Chart).

D-2.1.4 Data

Objective. To determine SUT's ability to transport DSN data traffic via a 2-wire analog interface.

Criteria. The SUT shall be capable of transporting the following data types:

a. Asynchronous modem traffic shall be able to be transferred at a minimum rate of 4.8 kbps.

b. Secure data via STE/STU. Shall be capable of passing:

1) 9600 kbps data speeds for STU-to-STU.

2) 9600 kbps data speeds for STU-to-STE.

3) 19.2 kbps for STE-to-STE calls.

Test Procedures. See detailed test procedures in section E-2.1.4 of appendix E.

Data Required. For data tests the following data shall be recorded:

a. Completion rates.

b. Data transfer rates.

c. Pattern slips.

d. Pattern losses.

e. Number of bit errors.

D-2.1.5 Video Teleconferencing (VTC)

VTC is not required of a 2-wire analog interface.

D-2.2 Integrated Services Digital Network (ISDN) Basic Rate Interface (BRI) National ISDN Standard 1/2 (NI 1/2) with T1.619a Line Interface

[Required: MFS, EOS, SMEO, PBX1]

D-2.2.1 Access

Objective. To determine SUT's ability to meet GSCR line requirements for an ISDN BRI NI 1/2 (T1.619a) interface for: DN identification, line signaling, alerting signals and tones, WWNDP, and call processing.

Criteria

a. DN. SUT must be capable of automatically identifying the DN.

b. Line signaling. SUT shall support:

1) American National Standards institute (ANSI) T1.619a.

c. Alerting signals and tones. SUT shall support:

1) Ringing.

2) DSN Information Signals. Ringback precedence call, preemption tone, call waiting, conference disconnect tone, override tone, and camp on.

d. WWNDP. SUT shall meet the following WWNDP format:

1) DSN user dialing consisting of:

a) Access digit N, where N is any digit 2-9.

b) P or S digit, where P is any digit 0-4 and S is any digit 5-9.

c) Route code 1X, where X is any digit 0-9.

d) Area code KXX, where X is any digit 0-9.

e) Switch code KXX, where X is any digit 0-9.

        f) Line number XXXX, where X is any digit 0-9.

    2) 7-digit/10-digit intraswitch dialing.

    3) 911 Conflict Resolution.

e. SUT shall be capable of providing Call Treatments IAW GSCR Section 4.1.

    1) Origination busy treatment.

    2) Busy/Idle status treatment.

f. SUT shall support Class of Service IAW GCSR Section 4.1.6.

    1) SUT Shall provide a minimum of 256 classmarks.

g. SUT shall support following screening IAW GSCR Section 4.5.8:

    1) Zone restriction capacity.

    2) Access restriction.

    3) COS Screening.

    4) Zone Restriction

Test Procedures. See detailed test procedures in section E-2.2.1 of appendix E.

Data Required

a. SUT configuration.

b. Alerting tones/signals.

c. Dialed digits.

d. Call processing.

e. Call treatments.

D-2.2.2 Voice

Objective. To determine the SUT's capability to transport DSN Voice services, non-secure and secure, via an ISDN BRI NI 1/2 (T1.619a) interface.

Criteria

a. Voice quality measured from end instrument to end instrument across the SUT must have an MOS of 4.0 95% of the time for non-secure voice.

b. MLPP precedence levels, invocation and operation shall be IAW GSCR Sections 3. SUT shall meet T1.619-1992 and T1.619a-1994 standards.

c. SUT Shall support DSN announcements IAW GSCR Section 3.1.3

d. SUT shall support MLPP COI IAW GSCR Section 3.8.9

e. SUT shall support MLPP Precedence Treatment IAW GSCR Section 3.8.9.

f. SUT shall be able to complete Secure calls via a STU-III, STE, or SWT. Secure calls shall complete at a minimum of 4.8 kbps.

Test Procedures. See detailed test procedures in section E-2.2.2 of appendix E.

Data Required. During the test conduct collect the following data:

a. MOS.

b. Secure call completion rates.

c. Secure call data rates.

d. MLPP signaling tones

D-2.2.3 Facsimile

Objective. To determine SUT's ability to transport DSN facsimile traffic across an ISDN BRI NI 1/2 (T1.619a) interface.

Criteria. Facsimile traffic shall be able to be completed across the SUT with Group 3 facsimile rates of 14.4 kbps. Completed facsimiles shall meet IEEE Std. 167A-1995 facsimile readability requirements.

Test Procedures. See detailed test procedures in section E-2.2.3 of appendix E.

Data Required. Completed facsimile IEEE Std. 167A-1995 (Facsimile Test Chart).

D-2.2.4 Data

Objective. To determine SUT's ability to transport DSN data traffic via an ISDN BRI NI 1/2 (T1.619a) interface.

Criteria. The SUT shall be capable of transporting the following data types:

a. Synchronous 56 kbps switched data. End-to-end bit integrity; less than 1 error in 10^9 bits over a 9-hour period.

b. Synchronous 64 kbps switched data. End-to-end bit integrity; less than 1 error in 10^9 bits over a 9-hour period.

c. N x 56 synchronous. End-to-end bit integrity; less than 1 error in 10^9 bits over a 9-hour period.

d. N x 64 synchronous. End-to-end bit integrity; less than 1 error in 10^9 bits over a 9-hour period.

e. Secure data via STE/STU. Shall be capable of passing:

1) 9600 kbps data speeds for STU-to-STU.

2) 9600 kbps data speeds for STU-to-STE.

3) 19.2 kbps for STE-to-STE calls.

Test Procedures. See detailed test procedures in section E-2.2.4 of appendix E.

Data Required. For data tests the following data shall be recorded:

a. Completion rates.

b. Data transfer rates.

c. Pattern slips.

d. Pattern losses.

e. Number of bit errors.

D-2.2.5 VTC

Objective. To determine SUT's ability to transport DSN VTC via an ISDN BRI NI 1/2 (T1.619a) interface.

Criteria. The SUT shall be capable of transporting H.320 VTC services.

Test Procedures. See detailed test procedures in section E-2.2.5 of appendix E.

Data Required

a. VTC successful completions.

b. VTC data rates.

c. VTC subjective quality measurement.

D-2.3 T1 Signaling System 7 (SS7) with T1.619a Trunk Interface

[Required: TS, MFS, EOS]

D-2.3.1 Trunking

Objective. To determine the SUT's ability to meet GSCR trunking requirements for a T1 SS7 (T1.619a) interface for: framing, line coding, signaling, alarms, timing, WWNDP, routing, trunk groups, call processing, channel associated signaling (CAS) to common channel signaling (CCS) trunk interworking, Pulse Code Modulation (PCM)-24/PCM-30 interoperation, and tandem switching (TS & MFS only).

Criteria

a. Framing and Line Code. SUT shall support Extended Super Frame (ESF) 1.544 Megabits per second (Mbps) 24 8-bit words plus 1 framing bit (193 bits) in 125 microseconds. Line Coding. Bipolar 8-Zero Substitution (B8ZS).

b. SUT shall support ANSI T1.619, T1.619a Signaling.

c. SUT shall support following Alarms:

1) Local. Detect Carrier Group Alarms (CGA) and busy all outgoing associated trunks. Initiate a RED alarm on loss of frame for 2.5 ± 0.5 seconds. Send remote YELLOW within the ESF facility data link. Restore to service within 15 ± 5 seconds after valid signal restored.

2) Remote. Within 35-1000 milliseconds of detecting a YELLOW alarm release all associated connections and remove from service. Within 20-1000 milliseconds after connecting equipment removes YELLOW alarm, restore affected circuits.

3) Channel. Detect channel alarm (DS0) and remove trunk from service.

d. SUT shall support following WWNDP format:

1) DSN user dialing consisting of:

a) Access digit N, where N is any digit 2-9.

b) Precedence (P) or service (S) digit, where P is any digit 0-4 and S is any digit 5-9.

c) Route code 1X, where X is any digit 0-9.

d) Area code KXX, where X is any digit 0-9.

e) Switch code KXX, where X is any digit 0-9.

f) Line number XXXX, where X is any digit 0-9.

2) 7-digit/10-digit intraswitch dialing.

3) 911 Conflict Resolution.

e. SUT shall support following routing:

1) TS, MFS: Route to primary route and up to nine alternate routes. Each route shall support a minimum of four trunk groups.

2) EOS: Route to primary route and up to five alternate routes. Each route shall support a minimum of one trunk group per route and 96 trunk members per trunk group.

f. Trunk groups. SUT shall be capable of removing trunks from service (make busy) and returning trunks to service (Make idle) IAW GSCR Section 2.5.

g. SUT shall support CAS to CCS trunk interworking IAW GSCR Section 3, table 3-12 and 3-13.

h. SUT shall support PCM-24/PCM-30 interoperation A-law to Mu-law conversions.

i. [Required: MFS,EOS]. SUT shall support Direct Inward Dialing (DID) IAW GSCR Section 2.3.2 as follows:

1) With DID, the switch seizes a DID trunk and outpulses the station line number to the PBX. If the called station's line is idle and not restricted from receiving terminating calls, the PBX alerts the called station and returns audible ringing on the incoming connection.

2) If the called station's line is busy, the PBX returns busy tone. If the called station is restricted from receiving terminating calls, the PBX routes the incoming call to an announcement, reorder tone, or to the attendant.

Test Procedures. See detailed test procedures in section E-2.3.1 of appendix E.

Data Required. During the test conduct the following data will be collected:

a. Switch Framing.

b. Line Coding.

c. ESF Pulse Mask.

d. Signaling characteristics.

e. Alarm information.

f. Dialed digits.

g. Call processing information.

h. Trunk Group status.

i. Switch configuration.

D-2.3.2 Voice

Objective. To determine the SUT's capability to transport DSN Voice services, non-secure and secure, via a T1 SS7 interface.

Criteria

a. Voice quality measured from end instrument to end instrument across the SUT must have an MOS of 4.0 95% of the time for non-secure voice.

b. SUT shall support MLPP shall be IAW GSCR Sections 3.4.3 and 3.9. SUT shall meet T1.619-1992 and T1.619a-1994 standards for:

1) Preemption for Reuse (Answered call).

2) Preemption for Reuse (Unanswered call).

3) Preemption Not for Reuse (Answered call).

4) Preemption Not for Reuse (Unanswered call).

5) BPA.

c. SUT shall be able to complete secure calls via a STU-III, STE, or SWT. Secure calls shall complete at a minimum of 4.8 kbps.

Test Procedures. See detailed test procedures in section E-2.3.2 of appendix E.

Data Required. During the test conduct, the following data will be collected:

a. MOS.

b. Secure call completion rates.

c. Secure call data rates.

d. SS7 MLPP signaling messages.

e. Switch announcements.

D-2.3.3 Facsimile

Objective. To determine SUT's ability to transport DSN facsimile traffic across a T1 SS7 interface.

Criteria. Facsimile traffic shall be able to be completed across the SUT with Group 3 facsimile rates of 14.4 kbps. Completed facsimiles shall meet Institute of Electrical and Electronics Engineers (IEEE) Standard (Std.) 167A-1995 facsimile readability requirements.

Test Procedures. See detailed test procedures in table E-2.3.3

Data Required. Completed facsimile IEEE Std. 167A-1995 (Facsimile Test Chart).

D-2.3.4 Data

Objective. To determine SUT's ability to transport DSN data traffic via a T1 SS7 interface.

Criteria. The SUT shall be capable of transporting the following data types:

a. Asynchronous modem traffic shall be able to be transferred at a minimum rate of 4.8 kbps.

b. Synchronous 56 kbps switched data. End-to-end bit integrity; less than 1 error in 10^9 bits over a 9-hour period.

c. Synchronous 64 kbps switched data. End-to-end bit integrity; less than 1 error in 10^9 bits over a 9-hour period.

d. N x 56 synchronous. End-to-end bit integrity; less than 1 error in 10^9 bits over a 9-hour period.

e. N x 64 synchronous. End-to-end bit integrity; less than 1 error in 10^9 bits over a 9-hour period.

f. Secure data via STE/STU. Shall be capable of passing:

1) 9600 kbps data speeds for STU-to-STU.

2) 9600 kbps data speeds for STU-to-STE.

3) 19.2 kbps for STE-to-STE calls.

Test Procedures. See detailed test procedures in section E-2 of appendix E.

Data Required. For data tests, the following data shall be recorded:

a. Completion rates.

b. Data transfer rates.

c. Pattern slips.

d. Pattern losses.

e. Number of bit errors.

D-2.3.5 VTC

Objective. To determine SUT's ability to transport DSN VTC via an SS7 interface.

Criteria. The SUT shall be capable of transporting H.320 VTC services.

Test Procedures. See detailed test procedures in section E-2 of appendix E.

Data Required

a. VTC successful completions.

b. VTC data rates.

c. VTC subjective quality measurement.

D-2.4 E1 SS7 (Q.735.3) Trunk Interface

[Required: TS, MFS, EOS] (For Europe use only)

D-2.4.1 Trunking

Objective. To determine SUT's ability to meet GSCR trunking requirements for an E1 SS7 (Q.735.3) interface for: framing, line coding, signaling, alarms, timing, WWNDP, routing, trunk groups, call processing, CAS to CCS trunk interworking, PCM-24/PCM-30 interoperation and tandem switching (TS & MFS only).

Criteria.

a. Framing and Line Coding. SUT shall support framing IAW International Telecommunication Union – Telecommunication Standardization Sector (ITU-T) G.704 with line rate of 2.048 Mbps. Line Coding shall be High Density Bipolar Order 3 (HDB3).

b. SUT shall support ITU-T Q.735.3 signaling.

c. SUT shall support alarms as follows:

1) Local. Detect CGAs and busy all outgoing associated trunks. Initiate RED alarm on loss of frame for 4.5 ± 0.5 seconds. Send remote YELLOW within the ESF facility data link. Restore to service within 15 ± 5 seconds after valid signal restored.

2) Remote. Within 35-1000 milliseconds of detecting a YELLOW alarm release all associated connections and remove from service. Within 20-1000 milliseconds after connecting equipment removes YELLOW alarm, restore affected circuits.

3) Channel. Detect channel alarm (DS0) and remove trunk from service.

d. SUT shall support WWNDP as follows:

1) DSN user dialing consisting of:

a) Access digit N, where N is any digit 2-9.

b) Precedence (P) or service (S) digit, where P is any digit 0-4 and S is any digit 5-9.

c) Route code 1X, where X is any digit 0-9.

d) Area code KXX, where X is any digit 0-9.

e) Switch code KXX, where X is any digit 0-9.

f) Line number XXXX, where X is any digit 0-9.

2) 7-digit/10-digit intraswitch dialing.

3) 911 Conflict Resolution.

e. SUT shall support following routing:

1) [Required: TS, MFS]. Route to primary route and up to nine alternate routes. Each route shall support a minimum of four trunk groups.

2) [Required: EOS]. Route to primary route and up to five alternate routes. Each route shall support a minimum of one trunk group per route and 96 trunk members per trunk group.

f. SUT shall support trunk groups. SUT shall be capable of removing trunks from service (make busy) and returning trunks to service (Make idle).

g. SUT shall support CAS to CCS trunk interworking IAW GSCR Section 3, table 3-12 and 3-13.

h. SUT shall support PCM-24/PCM-30 interoperation A-law to Mu-law conversions.

i. SUT shall support DID.

Test Procedures. See detailed test procedures in table section E-2.4.1 of appendix E.

Data Required. During the test conduct collect the following data:

a. Switch Framing.

b. Line Coding.

c. ESF Pulse Mask.

d. Signaling characteristics.

e. Alarm information.

f. Dialed digits.

g. Trunk Group status.

h. Switch configuration.

D-2.4.2 Voice

Objective. To determine the SUT's capability to transport DSN Voice services, non-secure and secure, via an E1 SS7 (Q.735.3) interface.

Criteria

a. Voice quality measured from end instrument to end instrument across the SUT must have an MOS of 4.0 95% of the time for non-secure voice.

b. SUT shall support MLPP IAW GSCR Sections 3.4.3 and 3.9. SUT shall meet Q.735.3 standards for:

1) Preemption for Reuse (Answered call).

2) Preemption for Reuse (Unanswered call).

3) Preemption Not for Reuse (Answered call).

4) Preemption Not for Reuse (Unanswered call).

5) BPA.

c. SUT shall be able to complete secure calls via a STU-III, STE, or SWT. Secure calls shall complete at a minimum of 4.8 kbps.

Test Procedures. See detailed test procedures in section E-2.4.2 of appendix E.

Data Required. During the test conduct collect the following data:

a. MOS.

b. Secure call completion rates.

c. Secure call data rates.

d. SS7 MLPP signaling messages.

e. Switch announcements.

D-2.4.3 Facsimile

Objective. To determine SUT's ability to transport DSN facsimile traffic across an E1 SS7 (Q.735.3) interface.

Criteria. Facsimile traffic shall be able to be completed across the SUT with Group 3 facsimile rates of 14.4 kbps. Completed facsimiles shall meet IEEE Std. 167A-1995 facsimile readability requirements.

Test Procedures. See detailed test procedures in table E-2.4.3

Data Required. Completed facsimile IEEE Std. 167A-1995 (Facsimile Test Chart).

D-2.4.4 Data

Objective. To determine SUT's ability to transport DSN data, asynchronous and synchronous, traffic via an E1 SS7 (Q.735.3) interface.

Criteria. The SUT shall be capable of transporting the following data types:

a. Asynchronous modem traffic shall be able to be transferred at a minimum rate of 4.8 kbps.

b. Synchronous 56 kbps switched data. End-to-end bit integrity; less than 1 error in 10^9 bits over a 9-hour period.

c. Synchronous 64 kbps switched data. End-to-end bit integrity; less than 1 error in 10^9 bits over a 9-hour period.

d. N x 56 synchronous. End-to-end bit integrity; less than 1 error in 10^9 bits over a 9-hour period.

e. N x 64 synchronous. End-to-end bit integrity; Less than 1 error in 10^9 bits over a 9-hour period.

f. Secure data via STE/STU. Shall be capable of passing:

1) 9600 kbps data speeds for STU-to-STU.

2) 9600 kbps data speeds for STU-to-STE.

3) 19.2 kbps for STE-to-STE calls.

Test Procedures. See detailed test procedures in section E-2.4.4 of appendix E.

Data Required. For data tests the following data shall be recorded:

a. Completion rates.

b. Data transfer rates.

c. Pattern slips.

d. Pattern losses.

e. Number of bit errors.

D-2.4.5 VTC

Objective. To determine SUT's ability to transport DSN VTC via an E1 SS7 (Q.735.3) interface.

Criteria. The SUT shall be capable of transporting H.320 VTC services.

Test Procedures. See detailed test procedures in section E-2.4.5 of appendix E.

Data Required

a. VTC successful completions.

b. VTC data rates.

c. VTC subjective quality measurement.

D-2.5 T1 CAS (MFR1, DTMF, DP) Trunk Interfaces

T1 CAS MFR1 [Required: TS, MFS, EOS]
T1 CAS DTMF [Required: TS, MFS, EOS, SMEO]
T1 CAS DP [Required: TS, MFS, EOS, SMEO]

D-2.5.1 Trunking

Objective. To determine SUT's ability to meet GSCR trunking requirements for a T1 CAS interface for: framing, line coding, signaling, alarms, timing, WWNDP, DSN switch outpulsing digit formats, routing, trunk groups, call processing, CAS to CCS trunk interworking, PCM-24/PCM-30 interoperation and tandem switching (TS & MFS only).

Criteria

a. SUT shall support T1 CAS framing and line coding. Framing shall be ESF 1.544 Mbps 24 8-bit words plus 1 framing bit (193 bits) in 125 microseconds. Line Coding shall be B8ZS.

b. SUT shall support CAS signaling as follows:

1) Multi-Frequency Recommendation 1 (MFR1).

2) Dual Tone Multi-Frequency (DTMF).

3) Dial Pulse (DP).

4) Signaling combinations: DTMF 2 way, DP 2 way, DTMF in-DP out, DP in-DTMF out, and MFR1 2 way.

5) Reverse battery and immediate start.

6) Normal and abnormal wink.

7) Guard timing.

8) Satellite interface.

9) Reselect and retrial.

10) Unexpected stop.

c. SUT shall support alarms as follows:

1) Local. CGAs and busy all outgoing associated trunks. Initiate RED alarm on loss of frame for 2.5 $\pm$ 0.5 seconds; send remote YELLOW within the ESF facility data link. Restore to service within 15 $\pm$ 5 seconds after valid signal restored.

2) Remote. Within 35-1000 milliseconds of detecting a YELLOW alarm, release all associated connections and remove from service. Within 20-1000 milliseconds after connecting equipment removes YELLOW alarm, restore affected circuits.

3) Channel. Detect channel alarm (DS0) and remove trunk from service.

d. SUT shall support following WWNDP format:

1) DSN user dialing consisting of:

a) Access digit N, where N is any digit 2-9.

b) Precedence (P) or service (S) digit, where P is any digit 0-4 and S is any digit 5-9.

c) Route code 1X, where X is any digit 0-9.

d) Area code KXX, where X is any digit 0-9.

e) Switch code KXX, where X is any digit 0-9.

f) Line number XXXX, where X is any digit 0-9.

g) 7-digit/10-digit intraswitch dialing.

h) 911 Conflict Resolution.

e. SUT shall be capable of DSN outpulsing digits as follows:

1) MFR1 IAW GSCR table 4-9.

2) DTMF IAW GSCR table 4-10.

f. SUT shall support routing as follows:

1) TS, MFS: Route to primary route and up to nine alternate routes. Each route shall support a minimum of four trunk groups.

2) EOS: Route to primary route and up to five alternate routes. Each route shall support a minimum of one trunk group per route and 96 trunk members per trunk group.

g. SUT shall support trunk groups by removing trunks from service (make busy) and returning trunks to service (Make idle).

h. SUT shall support CAS to CCS trunk interworking IAW GSCR Section 3, tables 3-12 and 3-13.

i. SUT shall support PCM-24/PCM-30 interoperation A-law to Mu-law conversions.

j. SUT shall support DID.

Test Procedures. See detailed test procedures in section E-2.5.1 of appendix E.

Data Required

a. Switch Framing.

b. Line Coding.

c. ESF Pulse Mask.

d. Signaling characteristics.

e. Alarm information.

f. Dialed digits.

g. Trunk Group status.

h. Switch configuration.

D-2.5.2 Voice

Objective. To determine the SUT's capability to transport DSN Voice services, non-secure and secure, via a T1 CAS interface.

Criteria

a. Voice quality measured from end instrument to end instrument across the SUT must have an MOS of 4.0 95% of the time for non-secure voice.

b. SUT shall support MLPP IAW GSCR Section 3.4.1:

1) Preemption for Reuse (Answered call).

2) Preemption for Reuse (Unanswered call).

3) Preemption Not for Reuse (Answered call).

4) Preemption Not for Reuse (Unanswered call).

5) BPA.

c. SUT shall be able to complete secure calls via a STU-III, STE, or SWT. Secure calls shall complete at a minimum of 4.8 kbps.

Test Procedures. See detailed test procedures in section E-2.5.2 of appendix E.

Data Required

a. MOS.

b. Secure call completion rates.

c. Secure call data rates.

d. CAS MLPP signaling.

e. Switch announcements.

D-2.5.3 Facsimile

Objective. To determine SUT's ability to transport DSN facsimile traffic across a T1 CAS interface.

Criteria. Facsimile traffic shall be able to be completed across the SUT with Group 3 facsimile rates of 14.4 kbps. Completed facsimiles shall meet IEEE Std. 167A-1995 facsimile readability requirements.

Test Procedures. See detailed test procedures in section E-2.5.3 of appendix E.

Data Required. Completed facsimile IEEE Std. 167A-1995 (Facsimile Test Chart).

D-2.5.4 Data

Objective. To determine SUT's ability to transport DSN data traffic via a T1 CAS interface.

Criteria. The SUT shall be capable of transporting the following data types:

a. Asynchronous modem traffic shall be able to be transferred at a minimum rate of 4.8 kbps.

b. Synchronous 56k bps switched data. End-to-end bit integrity; less than 1 error in 10^9 bits over a 9-hour period.

c. Synchronous 64 kbps switched data. End-to-end bit integrity; less than 1 error in 10^9 bits over a 9-hour period.

d. N x 56 synchronous. End-to-end bit integrity; less than 1 error in 10^9 bits over a 9-hour period.

e. N x 64 synchronous. End-to-end bit integrity; Less than 1 error in 10^9 bits over a nine hour period.

f. Secure data via STE/STU. Shall be capable of passing:

1) 9600 kbps data speeds for STU-to-STU.

2) 9600 kbps data speeds for STU-to-STE.

3) 19.2 kbps for STE-to-STE calls.

Test Procedures. See detailed test procedures in section E-2.5.4 of appendix E.

Data Required. For data tests the following data shall be recorded:

a. Completion rates.

b. Data transfer rates.

c. Pattern slips.

d. Pattern losses.

e. Number of bit errors.

D-2.5.5 VTC

Objective. To determine SUT's ability to transport DSN VTC via a T1 CAS interface.

Criteria. The SUT shall be capable of transporting H.320 VTC services.

Test Procedures. See detailed test procedures in section E-2.5.5 of appendix E.

Data Required

a. VTC successful completions.

b. VTC data rates.

c. VTC subjective quality measurement.

D-2.6 E1 CAS (MFR1, DTMF, DP) Trunk Interfaces

E1 CAS MFR1 [Required: TS, MFS, EOS, SMEO] (Required for Europe only)
E1 CAS DTMF [Required: TS, MFS, EOS, SMEO] (Required for Europe only)
E1 CAS DP [Required: TS, MFS, EOS, SMEO] (Required for Europe only)

D-2.6.1 Trunking

Objective. To determine SUT's ability to meet GSCR trunking requirements for an E1 CAS interface for: framing, line coding, signaling, alarms, timing, WWNDP, DSN switch outpulsing digit formats, routing, trunk groups, call processing, CAS to CCS trunk interworking, PCM-24/PCM-30 interoperation and tandem switching (TS & MFS only).

Criteria

a. SUT shall support E1 CAS framing and line Coding. Framing shall be IAW ITU-T G.704 with line rate of 2.048 Mbps. Line Coding shall be HDB3.

b. SUT shall support following E1 CAS signaling types:

1) MFR1.

2) DTMF.

3) DP.

4) Signaling combinations: DTMF 2 way, DP 2 way, DTMF in-DP out; DP in-DTMF out, and MFR1 2 way.

5) Reverse battery and immediate start.

6) Abnormal operation.

7) Call for service timing and Guard timing.

8) Satellite interface.

9) Reselect and retrial.

10) Control signaling.

c. SUT shall support alarms as follows:

1) Local. Detect CGAs and busy all outgoing associated trunks. Initiate RED alarm on loss of frame for 4.5 $\pm$ 0.5 seconds; send remote YELLOW within the ESF facility data link. Restore to service within 15 $\pm$ 5 seconds after valid signal restored.

2) Remote. Within 35-1000 milliseconds of detecting a YELLOW alarm release all associated connections and remove from service. Within 20-1000 milliseconds after connecting equipment removes YELLOW alarm, restore affected circuits.

3) Channel. Detect channel alarm (DS0) and remove trunk from service.

d. SUT shall support following WWNDP format:

1) DSN user dialing consisting of:

a) Access digit N, where N is any digit 2-9.

b) Precedence (P) or service (S) digit, where P is any digit 0-4 and S is any digit 5-9.

c) Route code 1X, where X is any digit 0-9.

d) Area code KXX, where X is any digit 0-9.

e) Switch code KXX, where X is any digit 0-9.

f) Line number XXXX, where X is any digit 0-9.

g) 7-digit/10-digit intraswitch dialing.

h) 911 Conflict Resolution.

e. SUT shall support routing as follows:

1) [Required: TS, MFS]. Route to primary route and up to nine alternate routes. Each route shall support a minimum of four trunk groups.

2) [Required: EOS]. Route to primary route and up to five alternate routes. Each route shall support a minimum of one trunk group per route and 96 trunk members per trunk group.

f. SUT shall support trunk groups by being capable of removing trunks from service (make busy) and returning trunks to service (Make idle).

g. SUT shall support CAS to CCS trunk interworking IAW GSCR Section 3, tables 3-12 and 3-13.

h. SUT shall support PCM-24/PCM-30 interoperation A-law to Mu-law conversions.

i. SUT shall support DID IAW GSCR Section 2.3.2.

Test Procedures. See detailed test procedures in section E-2.6.1 of appendix E.

Data Required

a. Switch Framing.

b. Line Coding.

c. ESF Pulse Mask.

d. Signaling characteristics.

e. Alarm information.

f. Dialed digits.

g. Trunk Group status.

h. Switch configuration.

D-2.6.2 Voice

Objective. To determine the SUT's capability to transport DSN Voice services, non-secure and secure, via an E1 CAS interface.

Criteria

a. Voice quality measured from end instrument to end instrument across the SUT must have an MOS of 4.0 95% of the time for non-secure voice.

b. SUT shall support MLPP IAW GSCR Sections 3.4.3 and 3.9. SUT shall meet CAS standards for:

1) Preemption for Reuse (Answered call).

2) Preemption for Reuse (Unanswered call).

3) Preemption Not for Reuse (Answered call).

4) Preemption Not for Reuse (Unanswered call).

5) BPA.

c. SUT shall be able to complete secure calls via a STU-III, STE, or SWT. Secure calls shall complete at a minimum of 4.8 kbps.

Test Procedures. See detailed test procedures in section E-2.6.2 of appendix E.

Data Required

a. MOS.

b. Secure call completion rates.

c. Secure call data rates.

d. CAS MLPP signaling.

e. Switch announcements.

D-2.6.3 Facsimile

Objective. To determine SUT's ability to transport DSN facsimile traffic across an E1 CAS interface.

Criteria. Facsimile traffic shall be able to be completed across the SUT with Group 3 facsimile rates of 14.4 kbps. Completed facsimiles shall meet IEEE Std. 167A-1995, facsimile readability requirements.

Test Procedures. See detailed test procedures in section E-2.6.3 of appendix E.

Data Required. Completed facsimile IEEE Std. 167A-1995 (Facsimile Test Chart).

D-2.6.4 Data

Objective. To determine SUT's ability to transport DSN data traffic via an E1 CAS interface.

Criteria. The SUT shall be capable of transporting the following data types:

a. Asynchronous modem traffic shall be able to be transferred at a minimum rate of 4.8 kbps.

b. Synchronous 56k bps switched data. End-to-end bit integrity; less than 1 error in 10^9 bits over a 9-hour period.

c. Synchronous 64 kbps switched data. End-to-end bit integrity; less than 1 error in 10^9 bits over a 9-hour period.

d. N x 56 synchronous. End-to-end bit integrity; less than 1 error in 10^9 bits over a 9-hour period.

e. N x 64 synchronous. End-to-end bit integrity; less than 1 error in 10^9 bits over a 9-hour period.

f. Secure data via STE/STU. Shall be capable of passing:

1) 9600 kbps data speeds for STU-to-STU.

2) 9600 kbps data speeds for STU-to-STE.

3) 9.2 kbps for STE-to-STE calls.

Test Procedures. See detailed test procedures in section E-2.6.4 of appendix E.

Data Required. For data tests the following data shall be recorded:

a. Completion rates.

b. Data transfer rates.

c. Pattern slips.

d. Pattern losses.

e. Number of bit errors.

D-2.6.5 VTC

Objective. To determine SUT's ability to transport DSN VTC via an E1 CAS interface.

Criteria. The SUT shall be capable of transporting H.320 VTC services.

Test Procedures. See detailed test procedures in section E-2.6.5 of appendix E.

Data Required

a. VTC successful completions.

b. VTC data rates.

c. VTC subjective quality measurement.

D-2.7 T1 ISDN Primary Rate Interface (PRI) NI 1/2 (T1.619a) Trunk Interface

[Required: TS, MFS, EOS, SMEO, PBX1]

D-2.7.1 Trunking

Objective. To determine SUT's ability to meet GSCR trunking requirements for a T1 ISDN PRI NI 1/2 (T1.619a) interface for: framing, line coding, signaling, alarms, timing, WWNDP, routing, trunk groups, CAS to CCS trunk interworking, PCM-24/PCM-30 interoperation and tandem switching (TS & MFS only).

Criteria

a. SUT shall meet T1 PRI framing and line coding. Framing shall be ESF 1.544 Mbps 24 8-bit words plus 1 framing bit (193 bits) in 125 microseconds. Line Coding shall be B8ZS.

b. SUT shall support NI 1/2, ANSI T1.619, and T1.619a Signaling.

c. SUT shall support alarms as follows:

1) Local. Detect CGAs and busy all outgoing associated trunks. Initiate RED alarm on loss of frame for 2.5 ± 0.5 seconds; send remote YELLOW within the ESF facility data link. Restore to service within 15 ± 5 seconds after valid signal restored.

2) Remote. Within 35-1000 milliseconds of detecting a YELLOW alarm, release all associated connections and remove from service. Within 20-1000 milliseconds after connecting equipment removes YELLOW alarm, restore affected circuits.

3) Channel. Detect channel alarm (DS0) and remove trunk from service.

d. SUT shall support WWNDP format as follows:

1) DSN user dialing consisting of:

a) Access digit N, where N is any digit 2-9.

b) Precedence (P) or service (S) digit, where P is any digit 0-4 and S is any digit 5-9.

c) Route code 1X, where X is any digit 0-9.

d) Area code KXX, where X is any digit 0-9.

e) Switch code KXX, where X is any digit 0-9.

f) Line number XXXX, where X is any digit 0-9.

1) 7-digit/10-digit intraswitch dialing.

2) 911 Conflict Resolution.

e. SUT shall support routing as follows:

1) [Required: TS, MFS]. Route to primary route and up to nine alternate routes. Each route shall support a minimum of four trunk groups.

2) [Required: EOS]. Route to primary route and up to five alternate routes. Each route shall support a minimum of one trunk group per route and 96 trunk members per trunk group.

f. SUT shall support trunk groups by being capable of removing trunks from service (make busy) and returning trunks to service (Make idle).

g. SUT shall support CAS to CCS trunk interworking IAW GSCR Section 3, tables 3-12 and 3-13.

h. SUT shall support PCM-24/PCM-30 interoperation A-law to Mu-law conversions IAW GSCR Section 7.3.

i. SUT shall support DID IAW GSCR Section 2.3.2.

Test Procedures. See detailed test procedures in section E-2.7.1 of appendix E.

Data Required

a. Switch Framing.

b. Line Coding.

c. ESF Pulse Mask.

d. Signaling characteristics.

e. Alarm information.

f. Dialed digits.

g. Trunk Group status.

h. Switch configuration.

D-2.7.2 Voice

Objective. To determine the SUT's capability to transport DSN Voice services, non-secure and secure, via a T1 ISDN PRI NI 1/2 (T1.619a) interface.

Criteria.

a. Voice quality measured from end instrument to end instrument across the SUT must have an MOS of 4.0 95% of the time for non-secure voice.

b. SUT shall support T1 PRI MLPP IAW GSCR Section 3.4.2. SUT shall meet T1.619-1992 and T1.619a-1994 standards for:

1) Preemption for Reuse (Answered call).

2) Preemption for Reuse (Unanswered call).

3) Preemption Not for Reuse (Answered call).

4) Preemption Not for Reuse (Unanswered call).

5) BPA.

c. SUT shall be able to be complete secure calls via a STU-III, STE, or SWT. Secure calls shall complete at a minimum of 4.8 kbps.

Test Procedures. See detailed test procedures in section E-2.7.2 of appendix E.

Data Required

a. MOS.

b. Secure call completion rates.

c. Secure call data rates.

d. ISDN PRI T1.619a MLPP signaling messages.

e. Switch announcements.

D-2.7.3 Facsimile

Objective. To determine SUT's ability to transport DSN facsimile traffic across a T1 ISDN PRI NI 1/2 (T1.619a) interface.

Criteria. Facsimile traffic shall be able to be completed across the SUT with Group 3 facsimile rates of 14.4 kbps. Completed facsimiles shall meet IEEE Std. 167A-1995 facsimile readability requirements.

Test Procedures. See detailed test procedures in section E-2.7.3 of appendix E.

Data Required. Completed facsimile IEEE Std. 167A-1995 (Facsimile Test Chart).

D-2.7.4 Data

Objective. To determine SUT's ability to transport DSN data traffic via a T1 ISDN PRI NI 1/2 (T1.619a) interface.

Criteria. The SUT shall be capable of transporting the following data types:

a. Asynchronous modem traffic shall be able to be transferred at a minimum rate of 4.8 kbps.

b. Synchronous 56k bps switched data. End-to-end bit integrity; less than 1 error in 10^9 bits over a 9-hour period.

c. Synchronous 64 kbps switched data. End-to-end bit integrity; less than 1 error in 10^9 bits over a 9-hour period.

d. N x 56 synchronous. End-to-end bit integrity; less than 1 error in 10^9 bits over a 9-hour period.

e. N x 64 synchronous. End-to-end bit integrity; less than 1 error in 10^9 bits over a 9-hour period.

f. Secure data via STE/STU. Shall be capable of passing:

1) 9600 kbps data speeds for STU-to-STU.

2) 9600 kbps data speeds for STU-to-STE.

3) 19.2 kbps for STE-to-STE calls.

Test Procedures. See detailed test procedures in section E-2.7.4 of appendix E.

Data Required. For data tests the following data shall be recorded:

a. Completion rates.

b. Data transfer rates.

c. Pattern slips.

d. Pattern losses.

e. Number of bit errors.

D-2.7.5 VTC

Objective. To determine SUT's ability to transport DSN VTC via a T1 ISDN PRI NI 1/2 (T1.619a) interface.

Criteria. The SUT shall be capable of transporting H.320 VTC services.

Test Procedures. See detailed test procedures in section E-2.7.5 of appendix E.

Data Required

a. VTC successful completions.

b. VTC data rates.

c. VTC subjective quality measurement.

D-2.8 E1 ISDN PRI (Q.955.3) Trunk Interface

[Required: TS, MFS, EOS] (Required Europe only)

D-2.8.1 Trunking

Objective. To determine SUT's ability to meet GSCR trunking requirements for an E1 ISDN PRI (Q.955.3) interface for: framing, line coding, signaling, alarms, timing, WWNDP, DSN switch outpulsing digit formats, routing, trunk groups, CAS to CCS trunk interworking, PCM-24/PCM-30 interoperation and tandem switching (TS & MFS only).

Criteria

a. SUT shall support E1 PRI framing and line coding. Framing shall be IAW ITU-T G.704 with line rate of 2.048 Mbps. Line Coding shall be HDB3.

b. SUT shall support ITU-T Q.955.3 signaling.

c. SUT shall support alarms as follows:

1) Local. Detect CGAs and busy all outgoing associated trunks. Initiate RED alarm on loss of frame for 4.5 ± 0.5 seconds; send remote YELLOW within the ESF facility data link. Restore to service within 15 ± 5 seconds after valid signal restored.

2) Remote. Within 35-1000 milliseconds of detecting a YELLOW alarm, release all associated connections and remove from service. Within 20-1000 milliseconds after connecting equipment removes YELLOW alarm, restore affected circuits.

3) Channel. Detect channel alarm (DS0) and remove trunk from service.

d. SUT shall support WWNDP format as follows:

1) DSN user dialing consisting of:

a) Access digit N, where N is any digit 2-9.

b) Precedence (P) or service (S) digit, where P is any digit 0-4 and S is any digit 5-9.

c) Route code 1X, where X is any digit 0-9.

d) Area code KXX, where X is any digit 0-9.

e) Switch code KXX, where X is any digit 0-9.

f) Line number XXXX, where X is any digit 0-9.

2) 7-digit/10-digit intraswitch dialing.

3) 911 Conflict Resolution.

e. SUT shall support routing. SUT shall route to one primary route and up to nine alternate routes. Each route shall support a minimum of four trunk groups.

f. SUT shall support trunk groups by being capable of removing trunks from service (make busy) and returning trunks to service (Make idle).

g. SUT shall support CAS to CCS trunk interworking IAW GSCR Section 3, tables 3-12 and 3-13.

h. SUT shall support PCM-24/PCM-30 interoperation A-law to Mu-law conversions IAW GSCR Section 7-3.

i. SUT shall support DID IAW GSCR Section 2.3.2.

Test Procedures. See detailed test procedures in section E-2.8.1 of appendix E.

Data Required

a. Switch Framing.

b. Line Coding.

c. ESF Pulse Mask.

d. Signaling characteristics.

e. Alarm information.

f. Dialed digits.

g. Trunk Group status.

h. Switch configuration.

D-2.8.2 Voice

Objective. To determine the SUT's capability to transport DSN Voice services, non-secure and secure, via an E1 ISDN PRI (Q.955.3) interface.

Criteria

a. Voice quality measured from end instrument to end instrument across the SUT must have an MOS of 4.0 95% of the time for non-secure voice.

b. SUT shall support E1 PRI MLPP IAW GSCR Sections 3. SUT shall meet Q.955.3 standard for:

1) Preemption for Reuse (Answered call).

2) Preemption for Reuse (Unanswered call).

3) Preemption Not for Reuse (Answered call).

4) Preemption Not for Reuse (Unanswered call).

5) BPA.

c. SUT shall be able to be complete secure calls via a STU-III, STE, or SWT. Secure calls shall complete at a minimum of 4.8 kbps.

Test Procedures. See detailed test procedures in section E-2.8.2 of appendix E.

Data Required. During the test conduct collect the following data:

a. MOS.

b. Secure call completion rates.

c. Secure call data rates.

d. ISDN PRI MLPP signaling messages.

e. Switch announcements.

D-2.8.3 Facsimile

Objective. To determine SUT's ability to transport DSN facsimile traffic across an E1 ISDN PRI (Q.955.3) interface.

Criteria. Facsimile traffic shall be able to be completed across the SUT with Group 3 facsimile rates of 14.4 kbps. Completed facsimiles shall meet IEEE Std. 167A-1995 facsimile readability requirements.

Test Procedures. See detailed test procedures in table E-2.8.3

Data Required. Completed facsimile IEEE Std. 167A-1995 (Facsimile Test Chart).

D-2.8.4 Data

Objective. To determine SUT's ability to transport DSN data traffic via an E1 ISDN PRI (Q.955.3) interface.

Criteria. The SUT shall be capable of transporting the following data types:

a. Asynchronous modem traffic shall be able to be transferred at a minimum rate of 4.8 kbps.

b. Synchronous 56 kbps switched data. End-to-end bit integrity; less than 1 error in 10^9 bits over a 9-hour period.

c. Synchronous 64 kbps switched data. End-to-end bit integrity; less than 1 error in 10^9 bits over a 9-hour period.

d. N x 56 synchronous. End-to-end bit integrity; less than 1 error in 10^9 bits over a 9-hour period.

e. N x 64 synchronous. End-to-end bit integrity; less than 1 error in 10^9 bits over a 9-hour period.

f. Secure data via STE/STU. Shall be capable of passing:

1) 9600 kbps data speeds for STU-to-STU.

2) 9600 kbps data speeds for STU-to-STE.

3) 19.2 kbps for STE-to-STE calls.

Test Procedures. See detailed test procedures in section E-2.8.4 of appendix E.

Data Required. For data tests the following data shall be recorded:

a. Completion rates.

b. Data transfer rates.

c. Pattern slips.

d. Pattern losses.

e. Number of bit errors.

D-2.8.5 VTC

Objective. To determine SUT's ability to transport DSN VTC via an E1 ISDN PRI (Q.955.3) interface.

Criteria. The SUT shall be capable of transporting H.320 VTC services.

Test Procedures. See detailed test procedures in section E-2.8.5 of appendix E.

Data Required

a. VTC successful completions.

b. VTC data rates.

c. VTC subjective quality measurement.

D-3 FEATURES & CAPABILITIES

D-3.1 Common Features

Objective. To determine SUT's ability to meet GSCR features common to single users and business customers.

Criteria. The SUT shall be capable of performing the following features:

a. [Conditional]. Selective call rejection IAW GSCR Section 2.1.2.

b. [Conditional]. Denied originating service IAW GSCR Section 2.1.3.

c. [Required: MFS, EOS, SMEO]. Code restriction and diversion IAW GSCR Section 2.1.4.

Test Procedures. See detailed test procedures in section E-3.1 of appendix E.

Data Required

a. Feature capabilities.

D-3.2 Attendant Features

[Required: MFS, EOS]

Objective. To determine SUT's ability to meet GSCR features for attendant services IAW GSCR Section 2.2.

Criteria. The SUT shall be capable of performing the following pass/fail attendant features:

a. Initiate all levels of precedence calls (ROUTINE through FLASH-OVERRIDE)

b. Provide a visual display of the calling number, calls of service, and precedence level for incoming direct dialed calls and calls diverted to the attendant.

c. Override any class of service (calling area or precedence) of the calling party on a call-by-call basis.

d. Attendant shall have the capability to override a busy line condition. Connections to central offices shall be restricted from busy verification. Attendant can enter an existing busy line to inform user of an incoming call. Override tone provided to busy line; selected stations may be classmarked to deny attendant break-in.

e. Attendant shall have capability to deflect calls to a night service, a fixed or manually selected DN.

f. Attendant shall have the capability to automatically recall a call that terminated to a busy or unanswered station.

g. Attendant shall be able to place calls in a waiting queue. Calls shall be retrieved in order of precedence level. Calls in queue shall not be lost when console placed out of service or forwarded to night service.

Test Procedures. See detailed test procedures in section E-3.2 of appendix E.

Data Required

a. Precedence capabilities.

b. Visual display capabilities.

c. Override capabilities.

d. Night service capabilities.

e. Auto recall capabilities.

f. Queuing capabilities.

D-3.3 Public Safety Features

Objective: To determine the SUT's ability to meet GSCR Section 2.4 public safety feature requirements

Criteria. The SUT shall be capable of performing the following features:

a. [Conditional]. The SUT shall provide basic 911 emergency services.

b. [Required: MFS, EOS, SMEO]. The SUT shall be capable of tracing calls terminating to a specified DN. When activated, the originating DN or incoming trunk number, terminating DN, and time and date are printed for each call.

c. [Required: MFS, EOS, SMEO]. The SUT shall be able to trace outgoing calls to a specified DN. A printout of the originating DN, outgoing trunk number or terminating number, and time and date is generated for each call.

d. [Required: TS, MFS, EOS, SMEO]. The SUT shall be capable of tracing tandem calls. A printout of incoming trunk number and terminating DN number, and time and date shall be generated per call.

e. [Required: TS, MFS, EOS, SMEO]. The SUT shall be capable of tracing calls in progress. The SUT shall identify the originating DN or incoming trunk for a call in progress.

Test Procedures. See detailed test procedures in section E-3.3 of appendix E.

Data Required

a. 911 capabilities.

b. Call trace capabilities.

D-3.4 Preset Conferencing

[Required: TS, MFS, EOS]

Objective. To determine SUT's ability to meet GSCR Section 2.6 requirements for Preset conferencing.

Criteria. The SUT must be able to support the following preset conference features:

a. Conference bridges:

1) Support ten (10) separate bridges with each bridge having the capacity for one originator and twenty (20) conferees. Each bridge shall have the capability to function as the "Primary" or "Secondary" or "Alternate" bridge that can interconnect to other bridges. Each preset conference bridge shall be fully capable of MLPP access and control.

2) Assign not greater than twenty (20) switch address numbers per bridge. Such address numbers may be a combination of subscriber lines and other conference bridge

access. Preset conference network(s) that require more than twenty conferees shall use the cascading bridge method of expanding the number of conferees beyond twenty conferees.

3) Access to preset conference equipment shall be by means of one or more KXX codes assigned solely for this use. A preset conference shall be established as follows:

a) The originating office shall screen the unique KXX office code, and precedence level to protect against unauthorized preset conference usage, and, where authorized, attempts the connection, either locally or by way of Interswitch Trunks (ISTs). The KXX conference codes shall be translated as vacant codes on 10-digit calls.

b) The originator of a preset conference shall key the digits KXX-XXXX. This address may be preceded by:

i) A route code for choice of data-grade or voice-grade circuits, in the form 1X-KXXXXXX.

ii) Access digit and precedence digit for precedence level treatment, in the form 9PKXX- XXXX, where P is any digit 0-4.

iii) Both the route code and precedence level in the form of 9P-1X-KXX-XXXX.

c) Translation of the 7-digit address shall determine routing to the appropriate switching center to obtain conferencing equipment.

d) At the appropriate switching center, the received address shall be translated to determine the conference group and the desired list of conferee addresses. The number assignments shall be made in accordance with the DSN numbering plan, DISAC 310-255-1.

e) If a called conferee's telephone is not answered, automatic disconnect is to take place within an adjustable interval of 15 to 60 seconds after a bridge leg is first connected to the conferee line.

f) Originators of preset conferences shall have the capability of adding up to five non-programmed conferees (within the 21 conferees capability) to the conference by sequentially keying each add-on address and connecting the conferee to the bridge, by hook flashing, dialing the potential conference, and then hook flashing again when the next conferee answers.

b. Conference Notification Recorded Announcement. When the conference equipment receives the first off-hook supervisory signal from an answering conferee, conference notification recording shall be applied, and shall continue as an audible announcement to answering conferees and to the originator until all conferees answer. The conference notification recording shall automatically be removed 2 seconds after the last conferee answers, indicating, by such removal, that the conferees have all answered and that the conference is ready to begin. The originator shall have the ability to remove the conference notification recording and force the conference via line or trunk by depressing the "A" or "#" key on a DTMF instrument or a programmed feature key on ISDN or digital instruments. Forcing the conference, prematurely, shall not interfere with attempts to complete the connections to unanswered conferees, or to add-on new conferees. Where access to secondary and tertiary bridges is necessary in a conference, arrangements shall be made so that the conference notification recordings applied at each bridge are not superimposed. Each bridge shall generate a notification recording that is audible only to those conferees on that bridge. When all conferees on a bridge have answered, the conference notification recording shall be removed automatically from that bridge 2 seconds after the last conferee answers. When the conference notification recording is removed automatically from a bridge, the notification recording from the adjacent bridge, if still continuing, shall then become audible to the originator and to the conferees on the remaining bridge(s). When a conferee disconnects, a conference disconnect tone as described in GSCR table 5-5 shall be sent to originator and other conferees in the conference.

c. Automatic Retrial and Alternate Address. Off-hook supervision shall be returned to the originator from each bridge when all conferees have answered or when the originator has forced the conference prior to all conferees answering. If answer supervision is not returned from any conferee location, within an adjustable interval of 15 to 60 seconds, one automatic retrial shall be made to the primary conferee address. Conferees may be provided with alternate addresses that the switch shall try when the call fails to complete to the primary address. If a call to a primary address fails to complete within two trials, the call shall be directed to an alternate address, if provided, and two call attempts shall be made to the alternate address.

d. Bridge Release. The releasing of primary, secondary, and tertiary bridges shall occur as follows:

1) The primary bridge shall be released when on-hook supervision is received on the originating port of the primary bridge or on all of the other conference bridge ports.

2) If on-hook supervision is received on the originating port of secondary or tertiary bridges, all subsequent connections and equipment shall be released.

3) A conference bridge shall be released after all attempts at call completion are made and no answers are received on all ports.

4) Release of conference bridges shall be such that it shall be impossible for the bridges to become locked together.

e. Lost Connection. If a connection to a conferee is lost, due to disconnection or preemption, a distinctive disconnect signal, defined as a conference disconnect tone, shall be provided to the conference originator and all conferees. If the originator is lost or preempted, the bridge shall be held up long enough for a preempt tone to be given to all conferees IAW GSCR table 5-5.

f. Secondary Conferencing. The switch shall provide the capability of secondary conferencing, which is the ability to interconnect conference bridges located at separate DSN switches. When a conference is activated and two or more of the addressees require a secondary bridge, the address shall be processed in the normal manner and directed toward the office serving the secondary equipment. The conference equipment shall be designed so that it may be used alternatively for primary or secondary conferences. Identical operational features, such as application and removal of the conference notification recorded announcement, shall be provided for both primary and secondary conferences.

g. Address Translation. Translation of the 7-digit conference address shall be as follows. The switch shall have the capability to translate three digits of the switch code. The first two digits of the four-digit line number may be utilized to identify the switching center at which conferencing equipment is located. The four-digit line number shall be translated to indicate the particular preset conference arrangement.

Test Procedures. See detailed test procedures in section E-3.4 of appendix E.

Data Required

a. Number of bridges supported.

b. Conference origination capabilities.

c. Announcements.

d. Retrial and alternate address capabilities.

e. Release capabilities.

f. Lost connections.

g. Secondary conferencing capabilities.

h. Address translation capabilities.

D-3.5 Nailed-Up Connections

[Required: TS, EOS, MFS]

Objective. To determine the SUT's ability to meet GSCR Section 2.8 requirements for nailed-up connections.

Criteria. The SUT shall provide the following functionality for nailed-up connections:

a. It shall be possible to establish a nailed-up connection between any two like (i.e., PCM on line type) terminations on the SUT or between the SUT and another DSN switch.

b. The nailed-up connections shall be capable of providing a direct interface to PCM transmission facilities terminating directly on the SUT at the standard PCM-24 DS1 level of 1.544 Mbps and the PCM-30 level of 2.048 Mbps supporting both CAS and CCS.

c. Supervision received at one side of the nailed-up connection shall be repeated at the other end. This applies to analog-to-digital and digital-to-digital connections.

d. All nailed-up connections through the switch shall be monitored by the normal maintenance routines to ensure proper operating paths through the switch. In the event that a PCM switching network fault affects a nailed-up connection, the SUT shall automatically reconfigure the connection around the fault.

e. The SUT shall be capable of "nailing-up" at least 10% of its circuits.

f. Nailed-up connections shall not be preemptable.

Test Procedures. See detailed test procedures in section E-3.5 of appendix E.

Data Required

a. Nail up completion rates.

b. Types of nail-ups supported.

c. Supervision transparency.

d. SUT nail-up monitoring.

e. Number of nail-ups possible.

f. Nail-up preemptability.

D-3.6 Precedence Access Threshold (PAT)

[Conditional: MFS]

Objective. To determine the SUT's ability to meet GSCR Section 2.11 requirements for control of precedence calls.

Criteria. The SUT must be capable of performing the following functions for controlling precedence calls:

a. All access lines shall be capable of being classmarked for/not for PAT screening. The switch shall provide a minimum of seven (7) PAT mechanisms. Each PAT mechanism shall accommodate any number of the switch's terminations and any combination of those terminations. The switch shall be equipped, through classmarks, to assign any PAT mechanism to any switch termination or conversely, any switch termination to any PAT mechanism. It shall be possible to classmark any switch termination as subject to PAT mechanism restrictions or not subject to PAT mechanism restrictions.

b. Outgoing calls shall be screened by classmark prior to being presented to the appropriate PAT mechanism. The classmark screening shall determine whether or not the outgoing call shall be presented to the PAT mechanism based on a comparison of the dialed address with the calling user's class of service privileges including precedence, calling area, route code, zone restriction, and Community of Interest (COI) classmarks. Outgoing calls that successfully pass all tests shall be presented to the appropriate PAT mechanism. Calls that fail one or more of the tests shall be rejected and given the proper signal and/or announcement.

c. Each PAT mechanism shall functionally consist of a set of Precedence Level/Calling Area (PL/CA) thresholds, a summation-of-thresholds count, and a total-calls-in-progress counter. The total-calls-in-progress counter shall count all calls in progress that were permitted by the PAT mechanism. The summation-of-thresholds count shall register the sum of all threshold levels associated with the PAT mechanism. The PL/CA thresholds place limits on the number of simultaneous calls allowed of a given combination of precedence level and calling area. There shall be five (5) precedence levels (ROUTINE, PRIORITY, IMMEDIATE, FLASH, FLASH OVERRIDE) and five (5) progressively wider calling areas (A1, A2, A3, A4, and A5) established. Each PAT mechanism shall have twenty-five (25) PL/CA thresholds. Associated with each of these precedence level and calling area combinations shall be a threshold setting and a calls-in-progress counter that registers the current count of the calls in progress for that precedence level/calling area combination. Each PAT mechanism shall functionally consist of the following elements:

1) A set of 25 PL/CA thresholds.

2) A set of 25 calls-in-progress counters. Each calls-in-progress counter is associated with a PL/CA threshold.

3) A summation-of-thresholds count.

4) A total calls-in-progress counter.

5) The calls-in-progress counter associated with a PL/CA threshold registers the number of calls in progress for that PL/CA combination.

6) The summation-of-thresholds count shall register the sum of the number of simultaneous calls allowed for each PL/CA combinations. This corresponds to summing the PL/CA thresholds.

7) The total-calls-in-progress counter shall register the total number of calls in progress that were permitted by the PAT mechanism.

d. Each PAT mechanism shall set the limit on the number of simultaneous calls permitted at each precedence level and calling area combination, subject to permissible overflow conditions specified below. The simultaneous calls limitation process shall be as follows:

1) A call that is presented to the PAT mechanism shall initially be tested against the PL/CA threshold that has the same precedence level and calling area of the call. The call will be permitted if the threshold setting for that PL/CA is greater than the calls in progress count for that PL/CA. If the call is permitted, the calls-in-progress count for that PL/CA and the total-calls-in-progress counter shall be incremented.

2) If the threshold setting is equal to or less than the calls in progress count for that PL/CA, the call will be tested against the overflow process.

e. The overflow process shall be that if a call has not been permitted by initial simultaneous-calls-limitation testing, as specified above, the call will be permitted if the three following conditions are met:

1) The sum of the threshold settings of the PL/CAs with equal and higher precedence levels and equal and wider calling areas than the call is greater than the sum of the calls-in-progress counts of the PL/CAs with equal and higher precedence levels and equal and wider calling areas than the call; and

2) The summation-of-thresholds count is greater than the total-calls-in-progress count.

3) The PAT algorithm shall never allow any FLASH/FLASH OVERRIDE (F/FO) precedence call to be blocked, dropped, delayed, discarded, suppressed or otherwise mutilated and interfered with in favor of calls of lesser precedence.

4) When a call is permitted due to the overflow procedure, it shall increment the calls-in-progress counter of the PL/CA that has the same precedence level and calling area combination of the call. It shall also increment the total-calls-in-progress counter.

f. The appropriate calls-in-progress counter for the PL/CA assignment of the call and the total-calls-in-progress counter shall be decremented when any of the following occur:

1) A call permitted by the PAT mechanism is unable to find an idle or preemptable outgoing trunk during the routing process.

2) A call permitted by the PAT mechanism is preempted or is discontinued for any reason.

g. Calls that are permitted DSN access shall search the appropriate DSN trunk groups for idle or preemptable trunks in accordance with normal routing procedures. ROUTINE precedence calls that are denied DSN access by a PAT mechanism shall be provided 120 Impulses per Minute (IPM) busy tone. PRIORITY or higher precedence calls which are denied DSN access by a PAT mechanism shall be provided the Precedence Access Limitation (PALA) recorded announcement.

h. The SUT shall be equipped to provide queuing for the PAT mechanism on both an off-hook and on-hook basis. When off-hook queuing is activated, calls shall be held in queue for a minimum of thirty (30) seconds per interval. The holding time shall be programmable and composed of a minimum of five (5) intervals. When on-hook queuing is activated, calls shall be held in queue for a minimum of five (5) minutes per interval. The holding time shall be programmable and composed of a minimum of five (5) intervals. Automatic callback to the caller shall be initiated after both the PAT mechanism outlet and the needed trunk are available. The switch shall be equipped to permit enabling and disabling either off-hook or on-hook queuing for each PAT mechanism on a local or remote basis.

i. The attendant shall be allowed to upgrade the precedence level and calling area of any call on a per call basis only. After the attendant has authorized the higher precedence level and calling area, the call shall be routed through the PAT prior to accessing the DSN. The Call Detail Recording (CDR) function shall record the escalated precedence level and calling area granted by the attendant.

j. Three operational measurement registers shall be associated with each PL/CA threshold. The first register shall be a peg count of the number of calls offered to the PAT mechanism logic at that precedence level/calling area combination. The second register shall record the number of calls not permitted by the PAT mechanism at that precedence level/calling area threshold combination. The third register shall record the number of calls successfully processed through the PAT mechanism at that precedence level/calling area combination that fail to find an idle or preemptable outgoing trunk.

k. The SUT shall be able to establish and change precedence level and calling area threshold settings and calling area restrictions of each PAT mechanism by administrative commands and shall not require switch down time to accomplish. A software log shall be maintained of any changes to the threshold levels for a period of ninety (90) days. An entry shall be automatically made in the log whenever a threshold level or group of threshold levels is changed. Each entry shall record the date and time the threshold levels were changed, the identity of the terminal from which the change was made (with user ID), the threshold levels resulting from the change and identification of personnel that made the change.

Test Procedures. See detailed test procedures in section E-3.6 of appendix E.

Data Required

a. PAT mechanisms.

b. PAT screening.

c. PAT thresholds.

d. Number of simultaneous calls supported.

e. PAT overflows.

f. PAT queuing.

g. PAT attendant interaction.

h. Operational register measurements.

i. SUT PAT administrative commands.

D-3.7 DSN Hotline Service

[Required: MFS, EOS, SMEO]

Objective. To determine the SUT's ability to meet GSCR Section 2.12 requirements for hotline services.

Criteria. The SUT must be capable of providing the following hotline services:

a. The SUT shall provide the following DSN hotline services:

1) The SUT shall be restrict DSN hotline services from using the following:

a) Hold (line terminator denied to put call on "HOLD").
b) Three-Way Calling.

c) Normal Call Transfer.

d) Electronic Key Telephone System (EKTS).

e) Ad hoc Conferencing.

2) The SUT shall allow the user to automatically initiate a voice or circuit mode data call to a previously designated user by initiating a service request and/or to act as a receiver for hotline originated calls and/or calls from designated parties on a screening list.

3) The SUT shall allow normal analog and digital (ISDN and non-ISDN) users to place DSN hotline calls by either going off-hook or call setup request without providing, in the call setup request, the called party information required by the network to route the call. Instead, the information for the destination call party setup is stored in the network by prior subscription.

4) The SUT shall also allow hotline services terminating restrictions to be provided to ensure that certain voice and/or circuit mode data hotline subscribers only receive calls from other specified users. The hotline service shall be provided on a subscription basis to specific users and may be withdrawn for administrative reasons or at the request of the subscriber.

5) The SUT shall use the subscribed information (Precedence level with voice and/or circuit mode data) to process a hotline call and establish it according to the normal network procedures. The SUT shall also indicate and retain that the call is a Hotline originated call with the subscribed precedence and voice and/or circuit mode data.

b. The SUT shall provide Protected Hotline Calling. The user may receive Hotline calls only from a specified list of parties and may originate calls to a specified destination only, called the Designated Called Party (DCP).

c. The SUT shall provide Pair Protected Hotline Calling. Each of the two designated users can only call each other and no third party can call either.

d. The SUT shall provide Unprotected Hotline Receiving. The user may originate calls to a specified destination only (i.e., the DCP) but may receive calls from any party including the DCP.

e. The SUT shall provide Protected Hotline Receiving. Users may receive calls only from a list of parties but may originate calls to any party.

f. ISDN hotline shall be interoperable with analog, digital, and other BRI lines using the DSN WWDNP “route code” identifier of “*5” for voice and “*6” for circuit mode data calls. (* The user does not dial these route codes.) The switch shall automatically dial hotline calls when an off-hook condition occurs and out pulse the appropriate route digit (i.e., 5 or 6).

Test Procedures. See detailed test procedures in section E-3.7 of appendix E.

Data Required

a. Hotline restrictions.

b. Protected hotline capabilities.

c. Unprotected hotline capabilities.

d. Hotline WWNDP interaction.

D-3.8 Tandem Switching

[Required: TS, MFS]

Objective. To determine the SUT's ability to meet GSCR Section 8 requirements for tandem switching.

Criteria

a. MLPP tandeming.

b. Tandem Call Detail Record.

Test Procedures. See detailed test procedures in section E-3.8 of appendix E.

Data Required

a. MLPP completion rates.

b. Call Detail Records.

D-3.9 Network Management (NM)

D-3.9.1 Interfaces

[Required: TS, MFS, EOS, SMEO]

Objective. To determine the SUT's ability to meet GSCR Section 9 requirements for NM.
Criteria. DSN switching systems shall provide DSN NM data to the Advanced DSN Integrated Management Support System (ADIMSS) via one of the three following physical interfaces:

a. Ethernet (IEEE 802.3)/Transmission Control Protocol/Internet Protocol (TCP/IP).

b. Serial (EIA-232)/Asynchronous.

c. Serial/Synchronous (X.25 and/or BX.25 variant).

All data that is collected shall be accessible through these interfaces. For DSN NM purposes, the telephone switch must provide four separate data channels. They may be physically separate (e.g., four distinct physical interface points) or logically separate (e.g., four user sessions through a single Ethernet interface). The data may be sent in American Standard Code for Information Interchange (ASCII), binary, or hexadecimal data or ASCII text designed for screen/printer display.

Data channels shall be capable of providing:

a. Alarm/log data.

b. Performance data.

c. Accounting data.

d. Switch access.

Test Procedures. See detailed test procedures in section E-3.9 of appendix E.

Data Required. Type of interfaces supported.

D-3.9.2 Measurements and Data Generation

[Required: TS, MFS, EOS, SMEO]

Objective. To determine the SUT's ability to measure and generate data in IAW GSCR Section 9.2 NM requirements.

Criteria

a. 30 seconds (secs), 5 minutes (mins), 15 mins, 30 mins, and daily reporting. The SMEO is not required to meet the 30 sec and 5 min traffic reporting requirements and may meet 15, 30, or 60 mins and daily reporting.

b. [Required: TS, MFS, EOS]. DSN traffic measurements (IAW GR-478-CORE Section 3.1.1).

1) DSN Intra-system Precedence Calls - Peg Count.

2) DSN Originating Calls - Flash Peg Count.

3) DSN Originating Hotline Calls - Peg Count.

4) DSN Originating Calls - Immediate Peg Count.

5) DSN Originating Calls - Flash Override Peg Count.

6) DSN Originating Calls - Priority Peg Count.

7) DSN Terminating Precedence Calls - Peg Count.

8) DSN Successful Preemption - Primary Intertoll Trunks for Reuse Peg Count.

c. [Required: SMEO]. DSN traffic measurements.

1) Traffic NM in either Provide the required Traffic NM data in 5, 15, 30, or 60 mins increments.

2) Trunk Group Number.

3) Trunks Equipped.

4) Outgoing Attempts.

5) Outgoing Overflows.

6) Trunk Group Usage (in seconds or hundred call seconds (CCS)).

7) Total Trunk Group Usage or Outgoing and Incoming Trunk Group Usage.

8) Switch Database Access.

Test Procedures. See detailed test procedures in section E-3.9 of appendix E.

Data Required.

a. SUT traffic reports.

b. SUT peg counts.

c. Trunk group information.

d. Outgoing call information.

e. Switch database access.

D-3.9.3 Fault Management

[Required: TS, MFS, EOS, SMEO]

Objective. To determine the SUT's ability to conduct fault management IAW GSCR Section 9.3 requirements.

Criteria

a. Detect fault conditions and report in near real-time.

b. Alarms may be sent as Simple Network Management Protocol (SNMP) traps.

c. Alarm messages must be distinguishable from administrative log messages.

Test Procedures. See detailed test procedures in section E-3.9 of appendix E.

Data Required

a. Detected faults.

b. Alarms generated.

D-3.9.4 Configuration Management

[Required: TS, MFS, EOS, SMEO]

Objective. To determine the SUT's ability to conduct configuration management IAW GSCR Section 9.4 NM requirements.

Criteria

a. Information Base Query Functions.

b. Information Base Update Functions.

   1) Update Data Integrity and Validation Tests.

   2) Best-Effort Update Operations.

   3) Delayed Update Activation.

   4) Derivation of Read-Only Attribute Values.

c. ISDN Functions

   1) Single Logical Operations Functions.

   2) Customer Premises Memory Administration.

3) Parameter Downloading.

4) Input Message Sequencing.

Test Procedures. See detailed test procedures in section E-3.9 of appendix E.

Data Required

a. Information Base queries.

b. Information Base Update queries.

c. ISDN functions.

D-3.9.5 Accounting Management

[Required: MFS, EOS, SMEO]

Objective. To determine the SUT's ability to conduct accounting management IAW GSCR Section 9.5 NM requirements.

Criteria

a. DSN CDR fields.

1) Start date of call.

2) Start time of call.

3) Either (or both) elapsed time of call or stop time of call.

4) Calling number

5) Called number (all dialed digits).

6) Precedence level of call.

7) Outgoing trunk group of call.

8) Outgoing trunk member of call.

9) [Required: MFS, EOS]. Incoming trunk group of call.

10) [Required: MFS, EOS]. Incoming trunk group member of call.

11) [Required: MFS, EOS]. Call answered/unanswered indicator.

12) [Required: MFS, EOS]. Conference call indicator.

13) [Required: MFS, EOS]. Customer/Business group Identification.

b. [Required: MFS, EOS]. CDR retention of 5 days.

Test Procedures. See detailed test procedures in section E-3.9 of appendix E.

Data Required

a. CDR field elements.

D-3.9.6 Performance Management

[Required: MFS, EOS, SMEO as stipulated below]

Objective. To determine the SUT's ability to conduct performance management IAW GSCR Section 9.6 NM requirements.

Criteria

a. Performance measurement data is polled.

b. Data provided in 5 or 15 mins intervals for MFS and EOS; SMEO in 5, 15, 30, or 60 mins intervals.

1) Trunk group number.

2) Trunk group far end Common Language Location Identifier (CLLI) (optional for SMEO).

3) Trunks equipped.

4) Trunks in service.

5) Outgoing attempts.

6) Outgoing overflows.

7) Incoming attempts.

8) Trunk group usage (in seconds or CCS).

a) Total trunk group usage, or

b) Outgoing trunk group usage and incoming trunk group usage.

c. Data provided in 5, 15, 30, or 60 mins interval for MFS, EOS, and SMEO.

1) Maintenance usage (in seconds or CCS).

2) Tandem call count (optional for SMEO).

3) Glare (optional for SMEO).

4) Outgoing failures (optional for all).

5) Incoming failures (optional for all).

6) Out match fail (optional for all).

7) Trunk group preemption failure (optional for SMEO).

8) Incoming calls preempted (optional for SMEO).

9) Outgoing calls preempted (optional for SMEO).

10) Incoming precedence calls preempted (optional for SMEO).

11) Outgoing precedence call preempted (optional for SMEO).

Test Procedures. See detailed test procedures in section E-3.9 of appendix E.

Data Required

a. Trunk group number.

b. Trunk Group far end CLLI.

c. Trunks equipped.

d. Trunks in service.

e. Outgoing attempts.

f. Outgoing overflows.

g. Incoming attempts.

h. Trunk group usage (in seconds or CCS).

1) Total trunk group usage, or

2) Outgoing trunk group usage and incoming trunk group usage.

i. Maintenance usage (in seconds or CCS).

j. Tandem call count.

k. Glare.

l. Trunk group preemption failure.

m. Incoming calls preempted.

n. Outgoing calls preempted.

o. Incoming precedence calls preempted.

p. Outgoing precedence calls preempted.

D-3.9.7 NM Controls

[Required: TS, MFS, EOS]

Objective. To determine the SUT's ability to implement NM controls data in IAW GSCR NM requirements.

Criteria

a. Automatic controls

1) Automatic Congestion Control.

2) Trunk Reservation (TRE).

3) Selective Incoming Load Controls (SILC).

b. Overload controls

1) Essential Service Protection (ESP).

c. Manual Controls

1) Trunk Group Controls.

a) Cancel From (CANF).

b) Cancel to (CANT).

c) SKIP.

d) Reroute (RR)

i) Single-via-reroute.

ii) Multiple-via-reroute.

2) Code Controls. Code Gapping Control on 3 to 10 digit destination code.

3) Total Office Manual Control Removal – Request/Response.

d. Treatment Options of Calls that are terminated.

1) No circuit announcement (NCA).

2) Emergency Announcement 1 (EA1).

3) Emergency Announcement 2 (EA2).

Test Procedures. See detailed test procedures in section E-3.9 of appendix E.

Data Required

a. Switch configuration.

b. NM controls.

D-3.9.8 Remote Access

[Required: TS, MFS, EOS, SMEO]

Objective. To determine the SUT's ability to process received remote commands IAW GSCR section 9.8 NM requirements.

Criteria. The SUT must be able to:

a. Process commands received via switch access channel.

b. Commands entered via command prompt or switch provided user interface.

c. Network related commands per configuration management section.

Test Procedures. See detailed test procedures in section E-3.9 of appendix E.

Data Required. Commands able to be processed (pass/fail).

D-3.10 ISDN Services

[Conditional: TS, MFS, EOS, SMEO, PBX1, PBX2]

Objective. To determine the SUT's ability to meet GSCR Section 10 requirements for ISDN Services.

Criteria. The SUT must be able to support the following ISDN Service features:

a. EKTS IAW GSCR table 10-3.

    1) Multiple DNs per terminal.

    2) Analog member of an EKTS group.

    3) Multiple DN appearances per call.

    4) Appearance call handling.

    5) Hold/Retrieve.

    6) Bridging/DN-Bridging.

    7) Intercom calling.

    8) Membership in a multiline hunt group.

    9) Abbreviated and delayed ringing.

    10) Automatic and/or manual bridged call exclusion.

b. Multiline Hunt Group IAW GSCR table 10-6.

    1) Linear and/or circular hunting.

    2) Assignment of non-hunt DN to hunt terminals.

Test Procedures. See detailed test procedures in section E-3.10 of appendix E.

Data Required

a. EKTS features supported.

b. EKTS call completions.

c. Multiline Hunt Group features supported.

d. Multiline Hunt Group call completion rates.

e. SUT announcements.

f. Error logs (if generated).

D-3.11 Synchronization

[Required: TS, MFS, EOS, SMEO, PBX1, PBX2]

Objective. To determine the SUT's ability to meet GSCR Section 11 requirements for synchronization.

Criteria

a. [Required: TS, MFS, EOS]. SUT shall support the external timing mode criteria:

1) SUTs shall accept two timing references and shall select one of the references as the active reference. This source of timing shall be used to time all transmitted synchronous signals (see figure D-1 below).

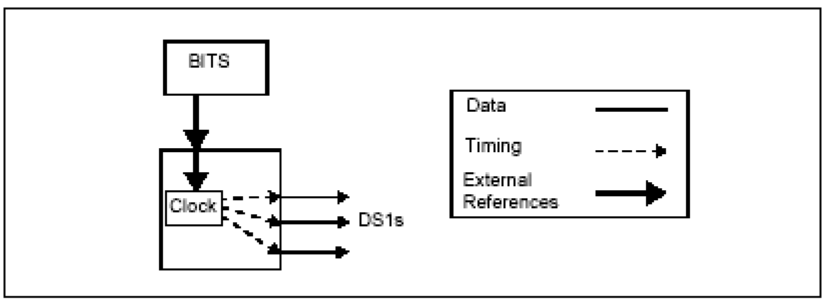

Figure D-1. External Timing Mode

2) SUT must be able to accept external Digital System Level 1 (DS1) reference signals in which the payload bits are not set to all-ones.

3) The DS1 external timing interface shall not carry traffic (although the DS1 reference signal may be a duplicate of a traffic-carrying signal).

b. [Required: TS, MFS, EOS, SMEO, PBX1]. SUT shall support line-timing modes as follows:

1) SUT shall have the capability to directly derive timing from a terminating synchronous signal. This source of timing shall be used to time all transmitted synchronous signals.

2) SUT shall provide the user the capability to provision any of its synchronous interfaces as a synchronization source.

3) With the exception of those containing stratum 4 clocks (SMEO and PBX1), SUTs that support line timing shall provide the capability for the user to provision more than one synchronous interface (if present) as a synchronization reference (e.g., one DS1 interface as “reference A” and another as “reference B”) (see figure D-2).

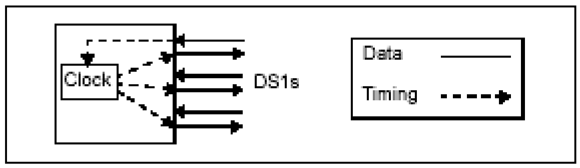

Figure D-2. Line Timing Mode

c. [Required: TS, MFS, EOS]. The SUT shall provide a stratum 3 or better internal clock.

d. [Required: SMEO, PBX1, PBX2]. The SUT shall provide a stratum 4 or better internal clock.

Test Procedures. See detailed test procedures in section E-3.11 of appendix E.

Data Required

a. Timing modes supported.

b. Frame slips.

c. Bit Error Rate (BER).

D-3.12 Reliability

[Required: TS, MFS, EOS, SMEO, PBX1]

Objective. To determine the SUT’s ability to meet GSCR Section 12 requirements for reliability.

Criteria

a. [Required: TS, MFS, EOS, SMEO].

b. [Required: PBX1]. System availability of 0.99997.

c. [Required: PBX1]. The SUT shall be able to detect, reconfigure, isolate, diagnose and report the location of single hardware failures.

Test Procedures. See detailed test procedures in section E-3.11 of appendix E.

Data required

a. Vendor method for calculating reliability.

D-3.13 Security

Objective. To determine the SUT's ability to meet GSCR Section 13 requirements for security.

Criteria. Criteria for security test procedures are contained within the Information Assurance Test Plan (IATP), a test plan developed to specifically address Department of Defense (DOD) security issues.

Test Procedures. Detailed test procedures for security are not addressed by this GSTP; rather, security test procedures are contained within the IATP.

D-4 REMOTE SWITCHING UNIT (RSU)

[Conditional: MFS, EOS, SMEO]

D-4.1 Normal Operating Condition

Objective. To determine the SUT's ability to meet RSU requirements IAW GSCR Section 2.10 for normal the normal operations.

Criteria. In normal operating condition, the SUT must be able to:

a. Provide the same user features as an EOS, SMEO, or PBX depending on how it is being used in the network. This means that the RSU must provide full DSN service as well as local intra-RSU service.

b. If used as an EOS, SUT must provide physical diverse routing to its associated host switch and have direct trunking to the Public Switched Telephone Network (PSTN).

c. Meet the normal operating condition requirements of Feature Specific Document (FSD) 30-23-0000, Issue 1, June 2000, which is a module of GR-532-CORE. SUT shall:

1) Provide the same feature complement as the host.

2) Provide individual and 2-party service, and Automatic Number Identification (ANI).

3) Provide Call treatments per GSCR Section 4.1.

4) Support class of service information. Lines served via an RSU should be easy to identify as such from the host.

5) Support the same or different NXX codes and/or Numbering Plan Area (NPA) codes.

6) Interpret and process 10-digit intra-office calls.

7) Provide RSU line screening and routing.

8) Provide overload controls to prevent overload of the RSU control complex and/or the host. It should be possible to invoke these controls independently for each RSU served by the host, whether or not the host or any other RSUs are in trouble. Host overload controls (protect the host processor, etc.), should not affect all the RSU-to-host channels from a given RSU so as to deny service to all customers on that RSU. However, it should be possible to limit the access of RSU customers when the host is in an overload condition.

9) Provide for both loop start and ground start line options.

10) Accept both DP and DTMF dialing.

11) Provide RSU NM measurements on an hourly basis:

a) Total originations from an RSU.

b) Total intra-RSU call attempts.

c) Total incoming call attempts to the RSU.

d) Total outgoing call attempts from the RSU.

e) Overflow counts.

f) Usage (equivalent to traffic minus maintenance usage).

g) Maintenance (out of service) usage.

12) Provide separate maintenance measurements. The separate measurements shall be included in the maintenance measurement report provided by the host. The host maintenance measurement report shall contain a separate section for each RSU served by the host.

13) Provide capability to detect and report alarms. Major equipment or service difficulties in an individual RSU should trigger an appropriate critical, major, or minor alarm at the host.

14) Support manually requested initiation of diagnostics.

Test Procedures. See detailed test procedures in section E-4.1 of appendix E.

Data Required

a. RSU-Host user features.

b. RSU routing and trunking capabilities.

c. FSD 30-23-0000 compliance.

D-4.2 Degraded Operating Conditions

Objective. To determine the SUT's ability to meet RSU requirements IAW GSCR Section 2.10 for degraded operations.

Criteria

a. Stand-Alone (SA)

1) SA operation IAW GR-532-CORE.

a) An interface to a special telephone access to the host from an inoperable RSU.

b) A visual alarm at the RSU to indicate that the RSU is in an SA mode of operation.

c) Provide status reports on alarms after exiting from SA.

d) All dialing patterns that the customer is accustomed to in normal operation (e.g., n-digit dialing, reverting calling) should be accepted in SA operation.

e) If it is not possible to process the customer's request (e.g., calls destined for lines not served by the RSU, or features requiring host resources not available at the RSU), then the customer should be routed to reorder or a recorded announcement (preferred).

f) During the transition to or from SA operation, it is required that all stable intra-RSU calls be maintained.

g) The SA RSU should continue to provide public telephone service, operation with key telephones, PBX lines, and pair gain devices terminated on its network, individual and/or multiple party ringing as provided normally.

2) Automated Message accounting not required.

3) Normal intraswitch MLPP.

b. Partial Stand-Alone (PSA)

1) PSA IAW GR-532-CORE

a) An interface to a special telephone access to the host or SCC from an inoperable RSU.

b) A visual alarm at the RSU to indicate that the RSU is in PSA mode of operation.

c) Provide status reports on alarms after exiting from PSA.

d) All dialing patterns that the customer is accustomed to in normal operation (e.g., n-digit dialing, reverting calling) should be accepted in PSA operation.

e) If it is not possible to process the customer's request (e.g., calls destined for lines not served by the RSU, or features requiring host resources not available at the RSU), then the customer should be routed to reorder or a recorded announcement (preferred).

f) During the transition to or from PSA operation, it is required that all stable intra-RSU calls be maintained, to minimize cut-off calls.

g) The PSA RSU should continue to provide public telephone service, operation with key telephones, PBX lines, and pair gain devices terminated on its network, individual and/or multiple party ringing as provided normally.

2) When the host switch to RSU links are saturated 3% of the user lines and umbilical line/trunk capacity shall be able to provide assured dial tone.

3) Normal intraswitch MLPP.

Test Procedures. See detailed test procedures in section E-4.2 of appendix E.

Data Required

a. Special interface to host.

b. Alarms.

c. RSU Status reports.

d. Dialing patterns.

e. Call completions/interruptions.

f. MLPP interactions.

D-5 VOICE OVER INTERNET PROTOCOL (VOIP)

[Conditional: MFS, EOS, SMEO, PBX1, PBX2]. If implemented the following paragraphs detail the specific requirements by switch type that VoIP systems must meet.

D-5.1 VoIP Systems

Objective. To determine the SUT's ability to meet the GSCR Appendix 3 requirements to process VoIP.

Criteria

a. [Required: TS, EOS, MFS, SMEO, PBX1, PBX2]. For VoIP Local Area Network (LAN) systems used in DSN, the voice quality shall have a MOS of 4.0 or better, as measured in accordance with Joint Technical Architecture (JTA) voice quality standards. This applies from handset to handset, and from handset to gateway trunk to the DSN.

b. [Required: TS, EOS, MFS, SMEO, PBX1, PBX2]. VoIP LAN system used in the DSN, the G.711 Pulse-Code Modulation (PCM) codec shall be used. No voice compression technique of any type shall be used within the DSN.

c. [Required: TS, EOS, MFS, SMEO, PBX1, PBX2]. VoIP systems shall be required to be certified and accredited in accordance with the Department of Defense Information Technology Security Certification and Accreditation Process (DITSCAP).

d. [Required: TS, EOS, MFS, SMEO - Conditional: PBX1, PBX2]. The VoIP system shall meet the NM functions described in Section 9 of this GSCR as applicable to switch type.

e. [Required: TS, EOS, MFS, SMEO, PBX1, PBX2]. The VoIP system shall support line timing modes as defined in GSCR, Section 11.1.1.2, Line Timing Mode.

f. [Required: TS, EOS, MFS - Conditional: SMEO, PBX1, PBX2]. The VoIP system shall provide internal clock requirements as described in the GSCR Section 11.

g. [Required: TS, EOS, MFS, SMEO, PBX1, PBX2]. The one-way system latency for the SUT/VoIP system shall be 60 milliseconds (msec) or less as averaged over any 5-minute period. The latency is to be measured from IP handset to egress from the SUT via a DSN trunk. An illustrative reference circuit diagram for this specification is shown in Figure D-3.

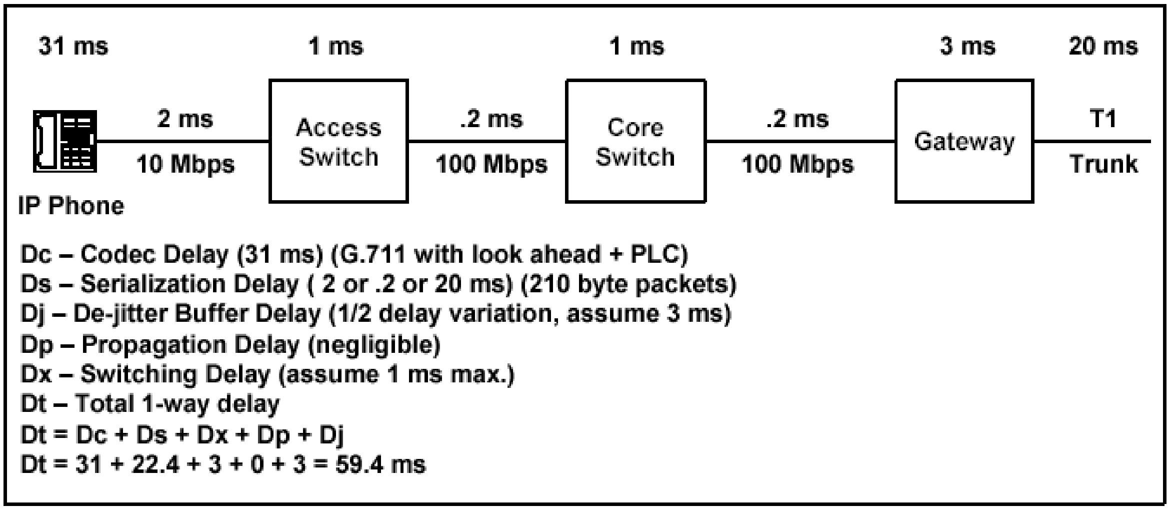

Figure D-3. LAN Delay from Mouth to Trunk to DSN Switch

h. [Required: TS, EOS, MFS, SMEO, PBX1, PBX2]. All IP devices in the VoIP system shall be IPv6 capable (in addition to maintaining interoperability with IPv4 systems) in accordance with the JTA.

Test Procedures. See detailed test procedures in section E-5 of appendix E.

Data Required

a. VoIP system diagram.

b. VoIP voice quality measurements.

c. VoIP system security profile.

d. VoIP system NM capabilities.

e. VoIP system timing capabilities.

f. VoIP system delay measurements.

g. VoIP system IPv6 supported.

D-5.2 LAN Requirements

Objective. To determine the ability of the Command and Control Voice Grade LAN (C2VGLAN) and Voice Grade LAN (VGLAN) to meet GSCR Appendix 3 requirements to support VoIP systems.

Criteria

a. [Required: C2VGLAN, VGLAN]. LAN Parameters:

1) The one-way packet delay for packets of an established call (signaling and media) within the LAN for a DSN VoIP system shall be 5 msec or less as averaged over any 5-minute period.

2) For voice media packets jitter shall be 5 msec or less as averaged over any five-minute period.

3) LANs shall be engineered for a theoretical packet loss of zero for voice packets; actual or measured voice packet loss within the LAN shall not exceed 0.05% averaged over any 5-minute period.

b. [Required: C2VGLAN (converged), VGLAN (converged) - Conditional: C2VGLAN (non-converged), VGLAN (non-converged)]. The routers and Ethernet switches shall support 802.1p to Differentiated Services (DiffServ) Code Point (DSCP) mapping and at least one of the following additional Class of Service (CoS) standards:

1) IEEE, 802.1p/Q.

2) DSCP.

3) Type of Service (TOS).

c. [Required: C2VGLAN (converged), VGLAN (converged). Conditional: C2VGLAN (non-converged), VGLAN (non-converged)]. The following is a listing of traffic streams prioritized from highest to lowest within the converged LAN (priorities shall be applied IAW the CoS models listed above):

1) Voice and video signaling and LAN NM (highest).

2) Voice and video media stream.

3) Data traffic (lowest)

d. [Required: C2VGLAN (converged), VGLAN (converged) - Conditional: C2VGLAN (non-converged), VGLAN (non-converged)]. The routers shall support at least one of the following queuing mechanisms:

1) Priority Queuing (PQ).

2) Custom Queuing (CQ).

3) Weighted Fair Queuing (WFQ).

4) Class-Based Weighted Fair Queuing (CBWFQ).

e. [Required: C2VGLAN (converged), VGLAN (converged) - Conditional: C2VGLAN (non-converged), VGLAN (non-converged)]. The routers shall support at least one of the following policing mechanisms:

1) DiffServ Per-Hop Behavior (PHB).

2) Generic Traffic Shaping (GTS).

3) Class-Based Shaping (CBS).

f. [Required: C2VGLAN (converged), VGLAN (converged) - Conditional: C2VGLAN (non-converged), VGLAN (non-converged)]. Ethernet switches shall support either Implicit or Explicit Virtual LAN (VLAN) membership for:

1) Port-based VLANs

2) Media Access Control (MAC) address-based VLANs

3) Layer 3 (or protocol)-based VLANs

g. [Required: C2VGLAN (converged), VGLAN (converged) - Conditional: C2VGLAN (non-converged), VGLAN (non-converged)]. NM and Voice traffic (signaling and media) shall be placed in a separate VLAN from data and video traffic. CoS and QoS measures shall be applied to the NM and the voice VLAN to guarantee bandwidth.

h. [Required: C2VGLAN, VGLAN]. All networks shall be designed not to exceed the IEEE recommended distances for Ethernet cabling.

i. [Required: C2VGLAN, VGLAN]. Ethernet switching platforms used in the LAN shall conform to the following IEEE standards:

1) 802.1d – Bridging.

2) 802.1p/Q – VLAN tagging.

3) 802.1s – Per-VLAN Group Spanning Tree Protocol.

4) 802.1v – VLAN classification by protocol and port.

5) 802.1w – Rapid Reconfiguration of Spanning Tree Protocol.

6) 802.1x – Port Based network access control.

7) 802.3ad – Link Aggregation Protocol.

j. [Required: C2VGLAN - Conditional: VGLAN]. The C2VGLAN shall have a hardware availability of .99999 (non-availability of no more than 5 minutes per year (mins/yr)). The vendor shall provide a reliability model for the system showing all calculations and showing how the overall availability will be met. The C2VGLAN shall have no single point of failure that can cause an outage of more than 64 telephony subscribers. In order to meet the availability requirements, all switching/routing platforms that offer service to more than 64 telephony subscribers shall have a modular chassis that provides at a minimum:

1) Dual power supplies.

2) Dual processors (control supervisors).

3) Termination Sparing.

4) Redundancy protocol

5) No Single Failure Point.

6) Switch Fabric or Backplane Redundancy.

k. [Required: C2VGLAN - Conditional: VGLAN]. In the event of a component failure in the network, all calls that are active shall not be disrupted (loss of existing connection requiring redialing) and the path through the network shall be restored within 2 seconds. All devices used to build redundancy shall be capable of handling the entire call processing load in the event that its counterpart device fails.

l. [Required: C2VGLAN - Conditional: VGLAN]. Each device in a LAN shall support the following NM features:

1) At least one of the following methods:

a) In-band. Ethernet Port. NM system connects to the network device using the same Ethernet network as the user traffic.

b) Out-of-band. NM system connects to the network device using a physically separated network from the network used for user traffic. This requires an additional network infrastructure to support management traffic.

i) Ethernet Port. NM system connects to the network device using a different Ethernet network than used for user traffic.

ii) Local port. Network management system connects to the network device via a local port (i.e., EIA-232 port) on the device using a computer, terminal or modem.

iii) Modem. Network managers connect to the network device remotely using an integrated modem interface on the device.

2) SNMP shall be used as a minimum for interfacing between the LAN devices and the NM system. In addition, if other methods are used for interfacing between LAN devices and the NM system, they shall be implemented in a secure manner such as with the following methods:

a) Secure Shell 2 (SSH2).

b) HyperText Transfer Protocol, Secure (HTTPS).

3) LAN devices shall be capable of measuring and reporting the following measurements:

a) Performance. The VoIP NM system shall collect statistics and monitor bandwidth utilization, delay, jitter, and packet loss.

b) Fault. The VoIP LAN transport elements (e.g., routers, Ethernet switches, etc.) shall provide status changes to the NM system by an automated capability in near real time (less than one minute) 99.95% (Objective) and 99.5% (Threshold) of the time. Sending SNMP satisfies the automated status change requirements or ASCII messages from the VoIP LAN components to the VoIP NM system.

c) Configuration. The VoIP NM system shall have the ability to perform remote configuration/reconfiguration of objects that have existing DOD JTA management capabilities or permit configuration/ reconfiguration via one of the protocols specified above.

4) For security and control purposes, VoIP LAN devices shall only be activated by a central control entity (e.g., help desk or service office). “Plug and Play” shall not be enabled. All DOD security policies and guidelines for IP networks shall be adhered to when designing the NM capabilities of the LAN.

m. [Required: C2VGLAN, VGLAN]. Bandwidth required per subscriber is 178.4 kbps (89.2 kbps each direction) for each IP call. This is based on G.711 with IP overhead (87.2 kbps) with VoIP signaling (2 kbps) included.

n. [Required: C2VGLAN, VGLAN]. LAN architectures proposed for VoIP systems shall be designed based on the following traffic engineering parameters:

1) Bandwidth for 64 subscribers is 5.71 Mbps (11.42 Mbps for two-way conversation); 64 subscribers can be serviced via a 10 Mbps full duplex link.

2) The physical interface between the LAN and Local Call controller/Gateway shall be a minimum of 100 Mbps Ethernet (full duplex).

3) Maximum number of IP telephony subscribers per 100 Mbps full duplex link to the IP Gateway or time Division Multiplexing (TDM) switch is 1024. For interface cards in the circuit switch or in the VoIP call control device that cannot support 1024 simultaneous IP subscribers, the number of subscribers is limited to the maximum number of simultaneous calls the interface card can support.

4) Access links (between IP phone/Ethernet switches) shall be switched 10 Mbps or 100 Mbps Ethernet.

o. [Required: C2VGLAN - Conditional: VGLAN]. For non-converged LANs, each switching/routing component's network links shall be engineered as follows:

1) 10 Mbps single link. Up to 64 IP telephony subscribers may be supported per 10 Mbps single link for the C2VGLAN. The VGLAN may support up to 100 IP telephony subscribers per 10 Mbps link because it does not have to support single point of failure requirements.

2) 10 Mbps link pair. Up to 100 IP telephony subscribers may be supported per 10 Mbps link for the C2VGLAN if reliability requirements are met. A 10 Mbps link pair is not required for the VGLAN.

3) 100 Mbps/1 Gbps single link. Up to 64 IP telephony subscribers may be supported per link for the C2VGLAN. For the VGLAN, up to 1024 IP telephony subscribers can be supported per link.

4) 100 Mbps/1 Gbps Link pair. Up to 1024 IP telephony subscribers can be supported per link pair on one device if reliability requirements are met.

p. [Required: C2VGLAN - Conditional: VGLAN]. For converged LANs, switching/routing component's network links shall be engineered as follows:

1) 100 Mbps/1 Gbps single link. Up to 64 IP telephony subscribers may be supported per link for the C2VGLAN. For the VGLAN, up to 256 for the 100 Mbps link and 1024 IP telephony subscribers can be supported per link.

2) 100 Mbps link pair. Up to 256 IP telephony subscribers may be supported per link pair for the C2VGLAN. Up to 512 IP telephony subscribers may be support per link pair for the VGLAN.

3) 1 Gbps link pair. Up to 1024 IP telephony subscribers may be supported per link pair for the C2VGLAN and VGLAN.

Test Procedures. See detailed test procedures in table E-5.

Data Required

a. LAN parameters: Delay, jitter, and packet loss.

b. LAN CoS, QoS, and policing mechanisms.

c. VLAN support.

d. IEEE standards supported.

e. Reliability.

f. NM capabilities.

g. Traffic engineering.

D-6 NETWORK INTERFACES

D-6.1 Public Switched Telephone Network (PSTN)

[Required: MFS, EOS, and SMEO]

Objective. To determine the SUT's ability to meet Chairman Joint Chiefs of Staff Instruction (CJCSI) 6215.01B requirements to gateway communications to the PSTN.

Criteria. The SUT must meet the following criteria:

a. Support a direct interface to a commercial interface, T1 ISDN PRI or T1 CAS.

b. Support positive identification control for the control of automatic interfaces that perform on-netting or off-netting.

Test Procedures. See detailed test procedures in section E-6.1 of appendix E.

Data Required

a. PSTN interfaces supported.

b. Identification control.

c. On-netting and Off-netting capabilities.

D-6.2 Tactical

Objective. To determine the SUT's ability to meet CJCSI 6215.01B requirements to gateway communications to the DOD tactical networks.

Criteria. The SUT must meet the following criteria:

a. Support a direct DSN interface to a tactical switch multiplex unit (SMU) at T1 or E1 CAS interface.

b. Support DSN to tactical voice and facsimile services.

Test Procedures. See detailed test procedures in section E-6.2 of appendix E.

Data Required

a. Interfaces supported.

b. Voice and facsimile services.

c. Voice quality measurements.

d. Facsimile quality measurements.

D-6.3 Defense Red Switch Network (DRSN).

Objective. To determine the SUT's ability to meet CJCSI 6215.01B requirements to gateway communications to the DRSN.

Criteria. The SUT must meet the following criteria:

a. Support a 2-wire analog access interface to the DRSN, a command and control element of the NCS.

b. Support DSN to DRSN voice services.

Test Procedures. See detailed test procedures in section E-6.3 of appendix E.

Data Required

a. Interface supported.

b. Voice services.

c. Voice quality measurements.

D-6.4 Enhanced Mobile Satellite System (EMSS)

Objective. To determine the SUT's ability to meet CJCSI 6215.01B requirements to gateway communications to the EMSS.

Criteria. TBD.

Test Procedures. TBD.

Data Required. TBD.

D-6.5 North Atlantic Treaty Organization (NATO) Gateway Communications System (NGCS)

Objective. To determine the SUT's ability to meet CJCSI 6215.01B requirements to interface to the NGCS.

Criteria. TBD.

Test Procedures. TBD.

Data Required. TBD.